IBA

TECHNICAL REVIEW

14

Latest Developments in Sound Broadcasting

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Contents

Introduction

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In 1972, when the ITA became the IBA and was appointed to provide a self-financing service of local radio broadcasting, our first problem was of finding channel space within what seemed a vastly overcrowded spectrum.

Also, we soon discovered that 'radio' presented much more of an engineering challenge than might have been expected in a year which marked the 50th anniversary of broadcasting in the United Kingdom. We found that the setting up of an entirely new system gave us a supreme opportunity of looking anew at the whole technical side of radio. Some of the engineering solutions — directional MF aerials, circular polarisation on VHF, compact studio centres — were described in *IBA Technical Review 5* published in 1974 (and now out of print).

By 1975 there were 19 ILR services on the air; but we then had to wait until 1978–79 before we were authorised to continue further towards our target of providing self-financing local radio to as much of the UK as possible.

In the intervening years, ILR has firmly proved itself as a new force in sound radio. The first local service solidly founded on VHF stereo, it has done much to awaken the public to the advantages of VHF/FM (even though the majority of people continue to listen, most of the time, on MF). It has already played a significant role in the investigation of 'surround-sound' broadcasting, resulting in the development of the 3-channel MSC system. As was originally intended, ILR programmes chiefly embrace topics of special interest to individual local communities, and include interchange of national news items through IRN. Our current plans include services for relatively small centres of population, and especially in areas remote from London and the south-east of England which, too long, have remained the fulcrum of British radio.

In recent years there has been a resurgence of interest whereby sound radio is re-emerging from the shadow of television as an important medium in its own right, unique in its appeal to the imagination and in its contributions of music. Everywhere there is a revival of interest and activity — with the public welcoming local broadcasting which has proved a thoroughly viable addition to national radio.

Technically, as this volume shows, there has been further progress in transmitter design; the exploitation of high efficiency 'Class D' techniques encouraging the introduction of all solid-state, modular MF transmitters with their promise of high reliability; switching-type encoders for stereo also mark the steady advance of digital-type systems in broadcasting. Similarly, the acoustics and design of compact production centres and self-drive studios are increasingly important considerations in the provision of local radio services to smaller and smaller communites.

It has been said sadly by Isaiah Berlin that 'the British are not much moved by economics, or economies, or by technology, and only a little by science . . . over the centuries they have adapted to the discoveries of science, but only for brief periods have they done so with enthusiasm, and the last time was more than a century ago, under the influence of a foreign prince'.

Evidence suggests that ILR has been an exception to the rule.

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The Acoustic Design of a Self-drive Studio

by C L S Gilford

Synopsis

This paper outlines the author's methods and rationale of
design of a speech studio forming part of a local radio
broadcasting station. The objects of design of any such
working area must primarily be functional and satisfaction of
performance specifications is of foremost importance. The
order of presentation follows closely that of the design process,
in which the requirements of insulation against existing
environmental noise, and against noise expected to be
generated within the building, are first considerations. The
second main section of the paper deals with the internal
acoustics of the studio. It stresses the need of applying, to the
design and distribution of the absorbers introduced to control
the reverberation, modern understanding of the diffusion of
the sound field.

INTRODUCTION

The two main objects in the acoustic design of any studio
or room where high-quality sound is to be originated or
assessed are:

(a) to ensure that the level of interfering noise at every
frequency is low enough to avoid degrading the quality of
the programme or to disturb the listeners;

(b) to achieve within the room, and on loudspeaker
reproductions of the programme, a sound quality which
is clear and free from distortions arising from sound
reflection within the room.

From the point of view of the designer, these aspects of
the design are generally referred to as sound insulation
and acoustics (or internal acoustics) respectively.

It is possible to use many different materials and
several approaches and, to this extent, design is a
personal matter. However, the IBA rightly insists on
proper standards of both background noise and acoustic
quality. Studios and control rooms are working tools;
and, as such, they should above all be functional. It is as
important to base a studio design on reliable methods and
well-tested materials to achieve the necessary standards as
it is with, say, a bridge. Some designers may prefer to use
materials with greater visual appeal or variety, but it is
usually at the expense of accurate knowledge of the
acoustic properties. The sound absorbers described in this
paper have been carefully researched over a long period,
and are of properties well understood; so that, if for archi-
tectural reasons any variation in design should be
necessary, the acoustic effect of it can be predicted.

Background noise and sound insulation will be first
considered, as in the logical order of design, and sections
on internal acoustics will follow.

STUDIO DIMENSIONS

The suitability of a site for a station will depend, among
other things, on the number of technical areas to be
accommodated within the available space, and on the
background noise levels. In general, it is cheaper to buy
an existing building for modification than to newly build.

3

The working area of a self-drive studio is largely for the engineering and programme experts to decide. In the UK, the average floor area of a self-drive studio is around 16 to 20 m². The gross area occupied in the building will be considerably greater, because the thickness of walls might need be as much as 600 mm to ensure adequate insulation, and allowance must be made for a sound lock at the entrance, as described later [see Sound Insulation para. (c)]. If music studios are to be included, further area allowance will be necessary to separate them from the rest of the studios by a corridor or a non-sensitive room.

The height of the studio is important and should be not less than 2·4 m. Adding the thickness of the floor slab, and a double ceiling with cavity deep enough to accommodate ducts, will require a minimum of nearly 3·2 m as the clear height of the ceiling of the original building.

BACKGROUND NOISE

(a) Acceptable Limits of Background Noise
It is not possible to specify an acceptable background noise level as a single weighted figure, as the reading on a sound level meter, because the noise normally present is spread over a wide frequency range, and an excessive noise energy over a small bandwidth could be very disturbing without very much affecting the weighted figure. Therefore, the acceptance limit must be specified as a graph of band level in octave bands against frequency, usually over the range 63 Hz to 4 kHz. These limits have varied widely between different authorities. The Institut fur Rundfunktechnik GmbH (IRT) based the curve for their studios on the self-noise of good condenser microphones, on the argument that there would be no practical method of detecting studio noise if it were below this. This criterion was below what was possible to hear on transmission and was therefore unnecessarily
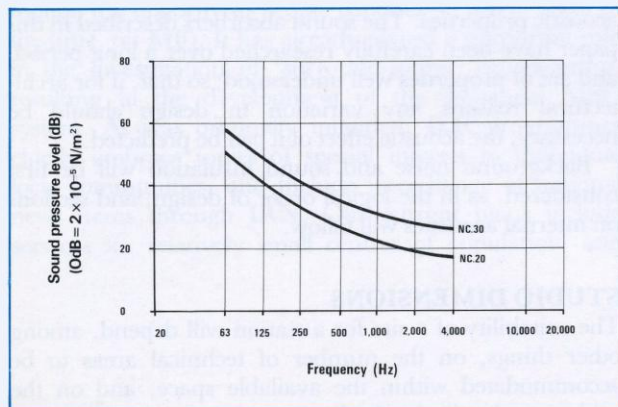
wasteful, and they now appear to be reverting to levels derived by other authorities from listening tests. The IBA uses the NC 20 criterion curve for studios and the NC 30 for control rooms (Fig. 1).

(b) Noise from Inside the Studio
Studio background noise can originate from inside the studio itself, from outside the studio and inside the building, and from outside the building. That from the studio itself consists of ventilation noise, the noise from fluorescent lights and their starters, from cooling fans in equipment, or tape recorder drives. These must be reduced at source and may be controlled by purchasing specifications or by making adjustments on site.

(c) Noise Originating from Outside the Studio
Noises from outside the studio can be controlled by sound insulation of the studio from other parts of the building. Those from outside the building are mostly due to aircraft, road and rail traffic, etc; and the levels at the outside of the building must be measured in octave bands in order to provide data for the insulation design.

Those from inside the building are of footfalls and conversation in the corridors, of lifts, toilet flushes and other plumbing, and of other devices not necessarily connected with the technical operations. In addition there are programme sounds and loudspeaker sound from the studios and control rooms. The levels of these cannot be measured before commencement of design and levels must be assumed, calculated on past experience or published information.

SOUND INSULATION

(a) Insulation from Noise External to the Studio
Vibration within the studio, arising from road traffic, can be reduced by introducing discontinuities between the road and foundations, such as covered trenches filled with loose material, by expansion joints in the base floor and filled with resilient material, and by floating the studio floor on a resilient blanket or on rubber mountings.

(b) Insulation from Other Technical Areas
A large body of experience has been built up from use of studio buildings, whereby the required sound insulation between any two areas can be specified from their respective functions.

The IBA lays down the requirements for the separation between these areas in the form of acceptance tests. A specified spectrum of sound is produced, having a flat portion up to 500 Hz and a decline of 2 dB per octave thereafter. The level of the flat portion of the test sound is suited to the purpose of the room into which it is radiated,
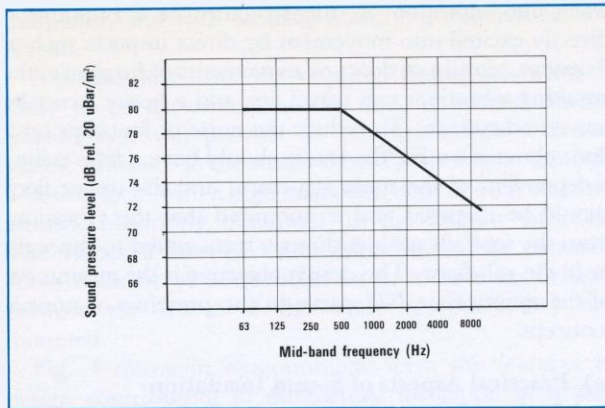

Fig. 1. Criterion curves for background noise in studios (NC20) and control rooms (NC30).

Fig. 2. '80dB' noise spectrum for sound insulation tests (applicable to control rooms and other areas using loudspeaker sound).

being 70 dB for speech-only studios, corridors, etc; 80 dB for rooms (such as control rooms) using monitoring or playback loudspeakers, and 100 dB for studios intended for the origination of all types of music.

For Production studios, and other studios intended for a variety of different sources of sound, the designer should discuss with the programme company and the IBA the exact purpose to which the studio will be put, and the maximum levels which are likely to be produced in it, in order to determine the source level for design of the insulation to protect the neighbouring studios. Fig. 2 shows the '80 dB' test spectrum.

In the acceptance tests, this spectrum is radiated into the 'source' room from loudspeakers, and the insulation must be such that the appropriate NC curve in the 'receiving' room is not exceeded. Direct subtraction of the NC level at each octave frequency from the specified source spectrum then gives the sound insulation to be achieved between the two areas.

At an early stage of the design of the station, the layout must be decided to arrange that, so far as possible, the studios are far away from external sources of sound, and from noisy equipment within the building. The insulation requirements of all partitions must be determined and the appropriate structures designed to give the needed insulation at each frequency with minimum expense and within minimum space.

(c) The Elements of Insulation

Sound can travel through a partition by two distinct mechanisms. In the first, the alternating pressures of the sound on the source side of the wall cause the wall to move bodily to and fro, as does the diaphragm of a loudspeaker, and the air at the other side is moved with the same amplitude as the movements of the wall, thus radiating sound into the receiving room. If, however, the

wall is porous or has open joints or cracks, air can leak through, setting into motion the air on the other side. This leakage of sound can be suppressed by covering the wall on one side with plaster or similar impervious material; but, if the leakage is allowed to remain, the sound insulation of the wall can be much reduced.

The sound insulating property of a partition is most conveniently defined as its Transmission Loss (TL), ie, the ratio, expressed in decibels, of the sound energy falling on the source side to that reaching the air on the receiving side. If the mass of single-leaf wall is doubled, the energy which reaches the other side is reduced to one quarter, because the amplitude of excitation is halved, and thus the RMS velocity is also halved. Hence, the kinetic energy of the excited air particles is reduced to one quarter. Also, if the frequency of the sound is doubled, the vibration velocity is halved, and with similar result. Thus, the transmission loss of an ideally heavy, limp single wall would be expected to increase by 6 dB for every doubling of the mass of the wall or of the frequency of the sound. In practice, doubling the mass increases the TL by rather less than 6 dB (say, 5 dB as a rough guide), while the frequency variation follows a general increase of 6 dB for every octave, with many large fluctuations in its course. To represent the TL of a partition by a single number, it is usual to quote the average TL in one-third octave intervals from 100 Hz to 3·2 kHz. There are other ways of expressing the TL of a wall as a single figure, particularly for certain applications such as the sound insulation of dwellings, but these need not here be described.

As an example, the mean TL of a 115 mm brick wall, plastered on one side, is 45 dB; a 225 mm wall, being twice as heavy, has a TL of 50 dB. Doubling the thickness again would increase the figure by only 5 dB, a very poor return for the extra mass. If the bricks for the 225 mm wall were instead used to build two 115 mm walls independent of one another, as for the sides of a wide corridor, each would contribute nearly its own 45 dB, giving a combined TL of nearly 90 dB. The shortfall is due to the interaction of the walls through the air, rather analogous to capacitive coupling between two elements of an electric circuit. If the distance between the walls is reduced, this coupling rises until, at nil separation, they behave as one. Therefore, to take advantage of double wall construction, there must be between them the largest possible gap; and all rigid wall ties, bridging by mortar waste and other solid connections, must be avoided.

In practice, by using a double wall instead of a single one of the same total mass, an advantage of about 5–15 dB can be gained within a practicable thickness. The design of highly insulating walls to take advantage of their
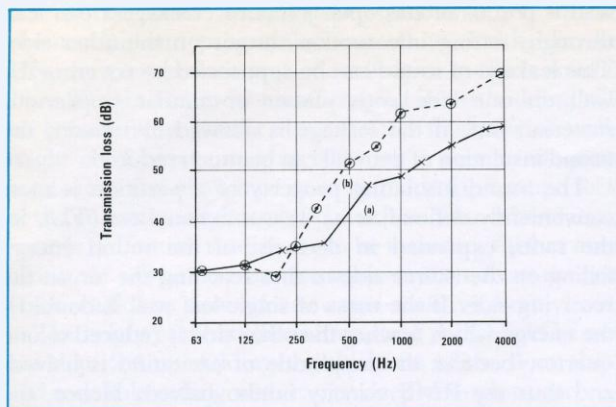
Fig. 3. Transmission loss curves for brick walls. a) 115 mm single wall; b) Two 115 mm leaves with 50 mm Cavity.

mass is not easy, because the behaviour of even a single leaf wall is less simple than as represented above. The stiffness of the wall combined with its mass causes resonances particularly at low frequencies. These greatly distort the ideal 6 dB/octave slope; and every partition has a so-called critical frequency at which there is a sharp dip in the TL curve.

The critical frequency is lowest in heavy, stiff walls, and highest in light, limp walls. One object of design is to reduce the severity of these dips and to bring them, if possible, to frequencies at which their presence is of less importance. With double or triple walls, there are additional severe resonances associated with the stiffnesses of the component leaves. Thus, the prediction of the exact transmission loss curve of a multiple leaf wall from the properties of its component parts is virtually impossible, so that recourse must be made to experience; or, to published information, where it exists. Fig. 3 shows the TL curves for a single 115 mm brick wall and two 115 mm leaves with 50 mm cavity.

Another factor to be taken into account is that of sound transmission occurring longitudinally in the structure of the building and giving rise to a parallel 'Flanking' propagation of sound round the edges of a partition wall. Flanking transmission, whether through the side walls or through accidental air paths, can limit the TL of an otherwise carefully designed wall. Therefore, the possibility of flanking must be considered at every stage of design. Generally speaking, structural paths can be interrupted by interposing resilient joints; and air paths such as joints, cracks, porosity, cable ducts, etc, can be given individual attention.

(d) Protection Against Impact Noise
The structure borne transmission mentioned above is originated by air waves in the source room setting the side

walls into vibration. If the structure of a building is directly excited into movement by direct impacts such as footsteps, closure of doors or movements of furniture, the resulting vibrations can travel far, and a heavy structure has no advantage. To reduce the noise of footsteps on a floor above a studio, the studio should have a false ceiling independent of the main structure, and the upper floor should be carpeted and so mounted that the vibrations from the footfalls are not directly transmitted to the walls or to the sub-floor. The design objective is the attainment of the appropriate NC curve in the presence of normal footsteps.

(e) Practical Aspects of Sound Insulation
Satisfaction of the acceptance tests for sound insulation between, say, a self-drive studio and an adjacent control room, requires a partition of slightly less than 50 dB mean isolation. This condition can be met by a well-designed double-leaf wall of which the internal wall of the studio is mounted on a resilient foundation and the floor of the studio is carried on a blanket of mineral wool. The mineral wool should be of a long-fibre type, and without the resin bonding which is used for many commercial wools. The object of the independent mounting of the inner leaf is to block structural transmission of sound from outside. It might be unnecessary in the control room which has a higher NC curve as its background noise requirement; but, in many cases, it will be desirable to use a triple wall partition, with the central leaf integral with the structure, but isolated from the floor slab above by a gap of about 13 mm packed with soft fibreboard or mastic. All walls should be plastered on their exposed faces, and brickwork must be carefully constructed with mortar joints completely filled.

Observation windows must be double or triple glazed, generally in accordance with the number of wall leaves, with substantial air spaces (100 mm upwards) between
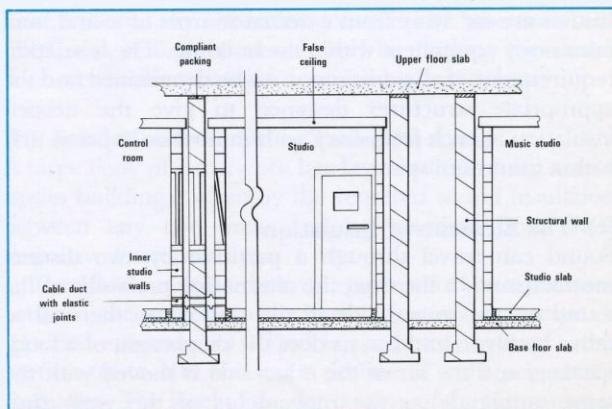

Fig. 4. Design features for good sound insulation.

the panes, and with acoustic absorbing material on the reveals.

A single door with the required transmission loss is likely to be excessively heavy; and, if opened during transmission will provide no protection from external noises. Entrance to the studio should therefore be through a 'sound lock' with a high insulation door at each end and acoustic absorbing treatment on the interior surfaces. The frames of the observation windows and the entrance doors must be so constructed as to include no bridges between the leaves of the walls in which they are mounted.

Fig. 4 shows in diagrammatic form the features in design contributing to satisfactory insulation of a self-drive studio from other technical areas.

INTERNAL ACOUSTICS OF THE STUDIO

(a) Reverberation

Concerning the internal acoustics of the studio, the two main factors to receive consideration are the total sound absorption by the surfaces of the studio with added absorbent treatment, and the correct distribution of the absorbing material over the six surfaces of the room.

The criterion specified first by the IBA for the acoustics is the reverberation time of the room, which is defined as the time taken for the sound level in the room to fall by 60 dB after a source of sound is switched off. When a sound is produced in any room, the sound waves travel from the source in all directions and are repeatedly reflected from the surfaces they encounter, gradually losing energy by absorption, until an equilibrium is reached between the energy supplied by the source and the decay of energy in the sound already in the room. When the source is switched off the sound decreases in level until it becomes inaudible. Too long a reverberation time in a speech studio results in undue prominence being given to sounds of certain frequencies.

Too short a reverberation time is not a disadvantage in a speech studio; but, because it requires a very large area of absorption, it is wasteful.

The reverberation time is measured by registering the decay of sound level on a chart recorder with a logarithmic scale. In ideal circumstances this produces a straight sloping graph from which the time for a fall of 30 to 35 dB is measured and the proportional time for 60 dB is calculated. This measurement is made at intervals of one-third of one octave from 63 Hz to 8 kHz. In practice, the decay curve usually has fairly large fluctuations which vary from one decay to the next, and it is necessary to estimate the average slope from a number of successive decays. If a pure tone is used as the test sound, the decays will vary greatly with very small changes in the frequency,

because the room has many natural resonances, following each other at small increments of frequency. Therefore, the test sound must be either a band filtered from random noise or a frequency-modulated tone, to give an averaging effect about the chosen test frequencies.

For small studios, the IBA specifies a maximum RT of 0·3s from 250 Hz upwards, with permitted increase at lower frequencies to a maximum of 0·44s at 63 Hz. As the size of the studio increases, the maximum permitted time increases slowly, but in such manner that the average absorption coefficient remains very nearly the same, irrespective of volume (within limits).

This is a useful provision of nature which enables the experienced acoustic designer to approximate more quickly to the final scheme of treatment.

Design for correct reverberation time consists of estimating the total absorption which will be present in the studio, including carpet, windows, doors, plaster surfaces, people etc. The required total absorption at each octave frequency is derived from the specified reverberation time by Eyring's formula (see below); and, from the two figures, the absorption to be added at each frequency is calculated.

(b) Absorption and Reverberation Time

The degree of absorption by any surface material is characterised by an absorption coefficient α, representing the ratio between the sound energy absorbed by the surface and the amount falling on it.

The coefficient varies with frequency and can take any value from almost zero in the case of a hard smooth surface such as ceramic tiles to as much as unity for specially developed materials. The absorption coefficients of the surfaces of a room determine the reverberation time (RT). If we put S as the total surface, and α the average absorption coefficient, the reverberation time is given by:

$$T = 0 \cdot 164 V / (-S \log \bar{\alpha}) \qquad (V = \text{Room volume}).$$

This is known as Eyring's formula. The original formula by Sabine, in which the denominator is simply $S\bar{\alpha}$, is inaccurate for small studios with short reverberation times such as those required for self-drive studios.

The absorption to be added is then assembled from areas of special acoustic absorbers covering the various ranges of frequency, of which the coefficients are known. The design process is simple in principle, but it is tedious because there is no practical alternative to trial and error, assisted by intuition and, at best, by a programmable calculator. It is best to start with low-frequency selective absorbers and to finish with the high-frequency end where adjustments are easy. The construction of selective absorbers is described below.

(c) Design of Absorbers

The most familiar type of sound absorber is the 'Acoustic Tile' which is widely used for noise reduction in offices, school halls, restaurants, and other places where people congregate. Such tiles are usually made of a mixture of plaster and mineral wool which produces a fissured surface, or of cane fibre with blind holes formed in it to assist the penetration of sound. In these porous or fibrous materials, the sound energy is dissipated by viscous friction as the air particles are driven to and fro by the alternating sound pressure. If the tile is stuck directly to a hard surface, only sound of short wavelength can effectively penetrate. The particle velocity is clearly zero where it meets the wall, and is maximum at a distance of a quarter of a wavelength from it, so that long wavelength sound never generates the high particle velocity necessary for high absorption within the material. Therefore, the performance of the tile is best at middle and high frequencies. The absorption at low frequencies can be improved by mounting the tiles on battens or on special suspension systems; but, at lower frequencies, the absorption is still not very good in relation to depth occupied.

Therefore, although absorbers such as those described above are useful to studio treatment, they must be supplemented by other absorbers with higher coefficients at middle and low frequencies. This is necessary because the absorption likely to be present in the untreated room is mainly at the higher frequencies, although some low frequency absorption occurs from vibration of the building structure.

Selective low-frequency absorption is best achieved by highly resonant structures which respond to the sound around a particular frequency, with a frictional or damping action which absorbs vibrational energy. Of these structures, the most efficient is the Helmholtz resonator, consisting of a volume of air connected by a narrow neck to the external sound field. An ordinary bottle is such a resonator; but, for efficiency at low frequencies, a large volume with a variable neck length is needed. Helmholtz resonators have frequently been used on the Continent, but they are very sensitive to positioning within the studio and need much special tuning and adjustment on site. Membrane absorbers, consisting of tuned loss inducing membranes stretched over a shallow cavity, were thoroughly researched in the post-war decade; but, whereas ordinary bituminous roofing felt has proved the best material for the membranes, it is nowadays compounded of synthetics which are very much less effective. Plywood panelling is a very effective bass absorber, acting over a frequency band wider than those of the specialised resonators. Also, deep pads of mineral wool are useful, provided that the neces-

sary space for these is available. Some designers have taken advantage of corners to accommodate such deep accumulations of mineral wool, so forming what are sometimes referred to as 'bass traps'. Only low-frequency absorbers can be confined to restricted locations in this way, because middle and high frequency absorbers need special positioning within the studio, as will be described in the next section.

A selective absorber for medium frequencies may consist of a blanket of mineral wool laid parallel to a hard surface, such as a studio wall, and fitted with a perforated cover of either hardboard or metal. In a simplified view of the action, the low-frequency cut-off is determined by the distance of the cover from the backing wall, as with the acoustic tiles described above. The upper cut-off is determined by the number and size of the perforations, ie, the percentage open area. On striking the cover, long wavelength sound is able to flow through the perforations; but, to short wavelength sound, the perforated cover acts as a reflector. A more accurate view, which enables prediction of the absorption/frequency curve, is that each perforation, in conjunction with the small 'cell' of air beneath it, acts as a miniature Helmholtz resonator. This has a broad peak of absorption which, by variation of the hole area and the distance from the backing wall, can be made to occur anywhere between 80 Hz and 1 kHz. In the case of the lower frequencies within that range, the performance can be improved by partitioning the air space with cardboard separators, as in a bottle crate, so that the air flow through the absorbing material is directed perpendicular to the backing wall. Fig. 5 shows the absorption
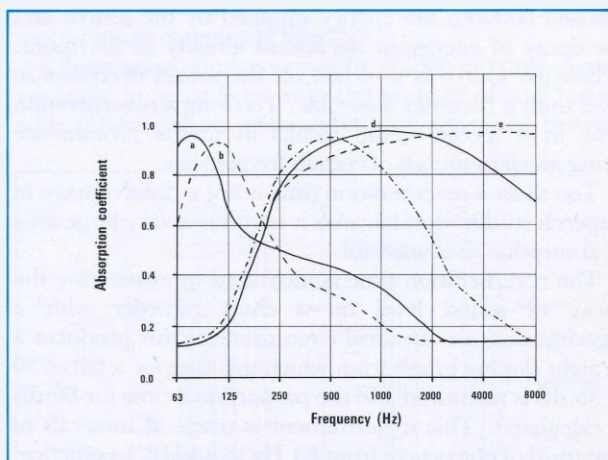

Fig. 5. Absorption characteristics of selective absorbers (average of collected data). a) membrane absorber; b) perforated panel absorber. Depth 180 mm, perforations with 0.5% open area. Partitioned air space; c) perforated panel absorber. Depth 50 mm. Open Area 5%; d) perforated panel absorber. Depth 50 mm. Open area 25%; e) As (d) but no cover.

characteristics of selective absorbers operating on various frequency bands.

A range of selective and broadband absorbers using the above principle of action, and in the form of modular boxes which can be fixed to ceilings or walls, is now obtainable from several British manufacturers.

(d) Distribution of Absorbers (Diffusion)

After deciding the construction and area of the absorbers required to give the desired absorption at all frequencies, the next consideration is of the way in which to distribute them on the available surfaces in order to ensure that the sound field is diffuse. The 'diffusion' of a sound field is the uniformity of distribution of sound energy and of particle velocity direction throughout the space. If the diffusion is poor, prediction of the reverberation time by Eyring's formula will be inaccurate. Poor diffusion becomes evident by long reverberation time in relation to the amount of absorption, irregular frequency response, strong standing wave systems giving rise to colourations on speech, large alterations in the slope of sound decay curves, flutter echoes and high-frequency ringing in the studio, all of these arising from programme sounds.

Quantification of the degree of diffusion by measurement or prediction is difficult and not fully understood. Despite great advances in research and in studio design throughout the last decade, the achievement of good diffusion and of accurately predicted reverberation time are matters of experience and judgement. To understand how the sound quality can be influenced by the arrangement of the absorbers, it must be realised that, when a sound is produced in the studio, it immediately sets up many standing wave systems of different frequency, formed by successive reflections from the wall surfaces. The most important of these systems are the simplest ones formed by repeated reflections between parallel pairs of walls. If all the absorbers were fixed on such a pair of walls, the standing waves between them would collapse rapidly, but those formed between other pairs of surfaces would be relatively unaffected. The decay curve would slope down rapidly at first as the absorbers removed the energy in the first set of standing waves, followed by a less sloping tail which, besides yielding a long RT in measurement would be subjectively long. If the absorbers were evenly distributed between all three pairs of surfaces, the rate of decay of all systems would be the same. Also, the decay curve would be straighter and ultimately shorter, and should give the predicted measured value. Therefore, the first rule in distributing the absorbers is to ensure that there is an adequate area of absorber at each frequency on each pair of surfaces. This is less important for the low-frequency absorbers, as mentioned above, and much of the needed absorbent material can conveni-
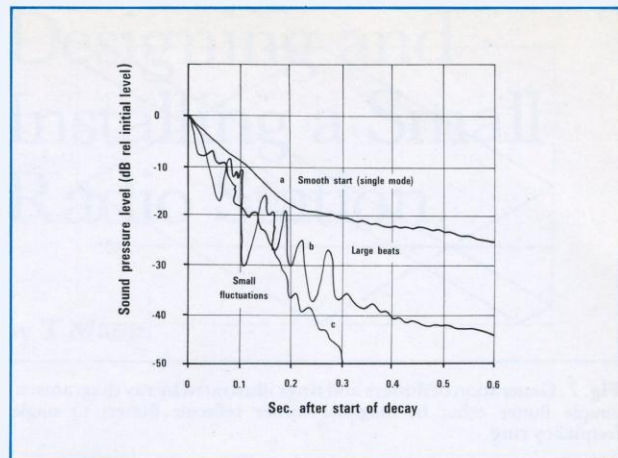

Fig. 6. Influence of absorber distribution on decay curves. a) All absorption on one parallel pair of walls; b) Same absorption on two pairs of walls; c) Same absorption distributed over all walls. (Diagrammatic.)

ently be mounted as a frieze or ceiling border. Fig. 6 shows how the sound decay curve changes as the absorption is spread over one, two and three wall pairs.

The next important point about distribution is that, if a sound wavefront is incident upon a flat reflecting wall, the wavefront reflected from the wall will be returned in the same shape. However, because of the ill effects of strong standing wave systems, it is desirable to scatter the wavefronts when they meet a surface.

Traditionally, this has been done by making the surface irregular in shape. To that end, many post-war studios have incorporated hemi-cylinders or even hemispheres stuck all over the walls. This is still a good method for larger studios and concert halls; but, if they are to be effective down to low speech frequencies, where colourations are easily excited in small studios, they should be at least 300 mm deep and so occupy a disproportionate amount of space in any small room. However, when sound falls on an efficient absorber, there is considerable scattering of sound which is not absorbed. This occurs mainly at the edges of the material, and equally so if the absorber is mounted flush with the rest of the wall. Thus, scattering is achieved at the frequency range covered by the absorber, and without any space penalty. As this is largely an edge effect, it is most marked if the large areas of absorber are sub-divided into small patches scattered in random fashion over the surfaces. Bad distribution of absorbers is also a cause of flutter echoes and of other faults in quality. If there is a serious shortage of middle and high frequency absorption on one pair of parallel walls, flutter echoes will be excited by any impulsive sounds such as dental consonants. If one wall pair is
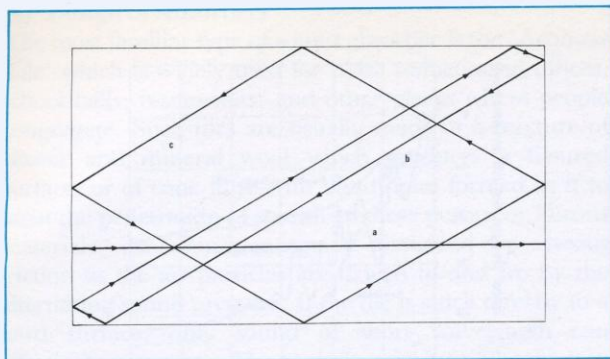
9

Fig. 7. Generation of flutters and rings illustrated by ray diagrams. a) simple flutter echo; b) diagonal corner reflector flutter; c) single-frequency ring.

adequately provided with absorbers but the others are more reflective, speech will set off ringing sounds at one or more spot frequencies. Flutter echoes can be caused by comparatively small areas of reflecting wall facing each other directly across the room; and so, wherever possible, the reflecting and absorbing areas should be positioned to avoid this effect. Small adjustments will usually remove flutters caused in this way; and it is desirable to use a modular arrangement for all the upper frequency absorbers so that minor rearrangements on test can easily be made. A rare but persistent type of flutter echo can occur if there are unbroken reflecting areas on each side of diagonally opposite corners of the room. Ray diagrams

for both forms of flutter and for single-frequency rings are shown in Fig. 7. To summarise, distribution of absorbers should be such as to equalise the absorption on the three pairs of room surfaces for each main frequency region, and to subdivide the absorbers for each range into small areas to promote diffusion. The arrangement of the individual areas on each wall should be carefully planned to prevent significant areas of reflecting surface from being placed exactly opposite each other.

DISCUSSION

The most important advance of the last few years in the understanding of the acoustics of small studios has undoubtedly been in the area of diffusion. Ten years ago there was no satisfactory method of measuring diffusion or of assigning a numerical value, and the relative importance and effects of directional distribution and edge diffraction could not be assessed. The position has much changed with the application of computer analysis to decay curves, by which numerical values can be assigned and the interaction of the above two factors clearly seen.

Apart from this, the last decade has been mainly one of slow development in design. The establishment of the local broadcasting stations, and the construction of large studio centres in the Middle East, have led to new appraisals of previous ideas and methods. Also, the strict standards of performance laid down by the IBA have helped to stimulate more accurate and economical engineering designs.

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He is married, has a daughter of pre-school age and lives in Cornwall.

Designing and Installing a Small Radio Station

by T Mason

Synopsis

In many respects the design of a small radio station is similar to that of a large one. The constraints of limited staff and budget make it important that the small station design must not only be right but be sufficiently flexible to adapt to changes in programme requirements without additional excessive expenditure. The technical design requirements depend on the type and variety of programmes planned.

This article describes the technical planning of such a station, certain hazards to be avoided, and the on-site duties of the station designer.

INTRODUCTION

The main problems in the work of designing and installing a small radio station usually arise from the need of cost saving. The total equipment budget for a small ILR station can be less than the cost of an OB/recording unit for a large station. For this reason the designer of a small ILR station must carefully consider the cost effectiveness of every item.

The programmes of a Local Radio Station must be designed to suit local requirements, tastes and interests. Therefore these stations differ enormously in their technical requirements. The first stage in planning is to translate the programme requirements into engineering hardware and studio layout. Almost all current and projected ILR stations are installed in converted premises; and so, the station designer has no standard design of studio layout from which to work. Instead, in the matter of premises, he must compromise where necessary to create a small radio station which may be quite different in layout from any other station. It must also be completed by an appointed date, and within a fixed budget, and will be required to operate within strict limits of performance.

SITING

Ideally, any radio station ought to be positioned at the centre of the area it is designed to serve. Among the factors governing the choice of the actual premises (assuming such choice to be available) there are noise factors to consider, such as the proximity of rail and road traffic, aircraft routes, buildings equipped with electric lifts, and other sources of audio and electrical interference. Availability of car parking, height of ceilings in the projected studio area to allow installation of air-conditioning and cabling, availability of main services such as gas, electricity, PO cables etc. and the projected ease, or otherwise, of obtaining planning permission are all factors which must be taken into account. At best, the choice of location will be a matter of compromise; in fact, the choice might already have been made before the station designer is appointed. These are the first influences on station design; and once these are settled the detailed studio layout can proceed.

STUDIO LAYOUT

Broadcast Studios in a small station are of two main types. Most music (disc) based programmes originate

11

from the type called 'self-drive control rooms'. In these, the presenter is responsible for the technical operation of the equipment. In small ILR stations this will include playing in music from disc, tape or cartridge, speech inserts off tape, either via PO line or live, and commercials. The presenter will also be expected to monitor transmission quality, to control the levels of transmission, operate phone-in equipment and check transmitter status alarms.

In a larger station much of the non-programme work, eg, overall level control, phone-in operation, playing in of commercials, may be handled by an engineer in a separate Master Control.

In a small station, the self-drive control room may include one or more interview microphone positions, and will probably have, or at least share, an associated studio. If the station is to provide a significant amount of speech programming a separate 'talks' area may be provided. In this second type of studio area the control room is occupied by an engineer/operator, possibly a producer, and even a telephone operator, while the programme presenter and guests sit in the associated studio. Ideally a third, non-broadcasting, studio/control room complex is provided for commercial production, live music recording and possibly even drama, quiz programmes etc. The particular requirements of this type of area render difficult the combination of its function with either type of broadcast area, although financial restraints might necessitate the adoption of a compromise talks/production design, or even a self-drive/production design. Another room might be equipped for 'phone-in' services, and there will be a racks room for the links for Outside Broadcasts (OBs), terminations for PO lines, clock and cue lights systems, etc. As examples of how the programme plans dictate the studio design let us consider the studio provision for two different stations, each serving a total audience of 500 000; the first, to serve five separate towns within an essentially industrial area, the second for a single residential area containing a more compact community. The first might require a predominance of music output, using two identical self-drive areas, a remote News studio in each town, OB links to the respective football grounds and conference centres, but relatively modest facilities for 'phone-in'. The second might require a larger speech content, could provide access facilities for local cultural groups producing their own programmes, coverage of local council affairs, and would probably need 'phone-in' arrangements with conference facilities.

Hence, the second station would be serviced with a single self-drive area, a 'talks' control room and studio, and comprehensive production facilities.

BUILDING OPERATIONS

When the essential studio layout has been finalised, the architect will translate these requirements into formal building plans including special acoustical treatment and acoustic isolation. The architect will require details of cabling, duct space, power sockets, and services such as clock and cue light wiring. In estimating the duct-work and cabling, due allowance should be made for future growth. The capital cost of providing excess duct and multicore cable (audio and control) is minute as compared with the cost of providing them at a later stage. Indeed, when the station becomes operational, any further work of cabling would certainly be very inconvenient and might even be impossible. The quantity surveyor will provide an estimate showing when each area will be likely to become ready for cabling and installation of equipment. In this, a generous time allowance for contingencies is desirable; because, of the many different materials required, short supply of any one material can hinder the whole job. Supplies of cable will be ordered accordingly.

DOCUMENTATION

A critical path system of progress recording will probably be too cumbersome to be effective; also, it might be insufficiently detailed to highlight problem areas. A loose-leaf diary should be prepared in advance, showing the work to be done in any week. The details will include the items of stores to be ordered, the building areas awaiting completion, and the allocation of personnel. It will provide a useful means of filing records or cable terminations, colour codings, and of rough, but accurate, sketches of particular circuit designs; and later, together with all details of the associated circuitry, the notes can be used in preparing full schematic diagrams of the wiring. Another advantage of such a diary is that it can easily be modified to suit changing circumstances.

EQUIPMENT

Orders for equipment, and for associated racking, can be placed as soon as the studio layout has been decided and the programme plans translated into hardware blocks. Equipment prices vary considerably; and, in a small station, a ruthless pruning of non-essential features can effect very large savings of capital expenditure. Typical items for elimination would be: unnecessary cabinet work, studio decor, and undue multiplicity of jackfields. For record-playback, use of 'fully professional' quality tape recorders will be required; but, for the purpose of playback-only (which can constitute a very large proportion of studio output) other tape recorders, of simpler design and much lower cost, will suffice. However, before

cheaper 'semi-professional' machines are purchased, the manufacturers' specification should be carefully examined, even if the machine is claimed to 'meet code of practice'.

The IBA Code of Practice figures for tape recorder performance relate, not to any individual machine, but to a recording made on one machine and then played back either on that machine or on another machine used for similar record/playback purposes. This requires a machine performance considerably better than the figures stated in the COP. Similarly, with Control Desks. The IBA Code of Practice relating to these applies, not to an individual desk alone, but to the desk performance via an equipment chain which includes microphone wiring, mixing desk, wiring to the central jackfield and, in certain instances, to distribution amplifiers.

Hence, any figures for equipment performance, as quoted in the IBA Code of Practice, should be used only as a starting point when specifying equipment for purchase. The Quality Control Section of the IBA have extensive experience in all engineering aspects of ILR, and will be able to offer constructive advice on a great variety of problems, from the measured performance of an individual tape machine, to the type of door seals likely to give best results. They should be kept informed at all stages of the studio design and layout.

PROVISION OF OTHER SERVICES

Provision of mains supplies, of water, gas and electricity, will be managed by the architect and his Clerk of Works; but that of Post Office communications is a responsibility of the station designer. The local Post Office Sales Department representatives will be able to assess the requirements of PBX, exchange lines, teleprinter service, key and lamp units etc., because these will be within their prior experience; but they might be less familiar with lines for OB services, or with lines and equipment for 'phone-in' services; and provision of these items can sometimes present difficulty. However, if the precise requirements of the station are explained, and good working relationship with the Post Office is maintained, most requirements will be met. Any case of difficulty should be referred to the IBA Lines Section who are the co-ordinating body for Post Office matters.

STATION COMPLETION

When the technical installation has been completed, a team of engineers from the IBA Quality Control Section will visit the station for the purpose of acceptance testing. This is to ensure that the acoustical and electrical performances of all the broadcasting areas meet the requirements of the IBA Code of Practice.

CONCLUSION

It is especially important that a small station, with limited staff and resources, shall operate as efficiently as possible. A large station might be able to afford an extra telephone operator, or an extensive re-engineering after a short while to change the studio layout and facilities to meet unforeseen programme plans; but a small station must be right from the start, and must be sufficiently flexible to adapt to future changes in the programme format. Although this article relates specifically to the designing and installing of a small ILR station in the UK, most of the information would be applicable to similar, and even larger, provision overseas.

PHILIP DARBY, CEng, MIERE, began his broadcasting career at the BBC where he held several engineering posts. On joining the IBA in 1955 he immediately wrote himself into broadcasting history by completing the log as Senior Shift Engineer on duty at Croydon transmitting station on the first night of ITV. Subsequently he was at Emley Moor and Engineer-in-Charge of Dover transmitting station. He has been Head of the IBA Quality Control Section since its formation in 1967 until 1981. He is married and lives in Hampshire.

Aspects of Measurement and Operation

by P J Darby

Synopsis

The responsibility for routine technical quality assessment has been delegated by the IBA to the Independent Local Radio companies. Therefore it is essential for IBA Quality Control Section to maintain close liaison with the programme company engineers.

This article describes the reasons for and methods of applying Codes of Practice tests at ILR studio centres. The author provides a list of the main items of test equipment necessary for making the electronic measurements and includes notes concerning the special requirements of each equipment. The possibility of introducing new equipment to improve the accuracy of assessment and to speed up test procedures is discussed.

INTRODUCTION

The Independent Broadcasting Authority Act of 1973 requires the Authority to maintain good technical standards in both the Independent Television and Independent Local Radio services. In the case of the television service, all programmes are assessed for technical quality by IBA engineers in rooms specially designed to provide professional monitoring conditions. Technical defects on any programme are discussed by telephone with the programme company engineers, and all significant problems receive immediate attention. Reports on the technical quality of all programmes are sent by the monitoring stations to Quality Control Section who issue a weekly analysis in respect of each company's output. They also maintain complete records and initiate remedial action as necessary to ensure that appropriate standards are maintained.

A large amount of programme material is pre-viewed by IBA Quality Control staff before being included in the transmission schedules.

Prior to the start of the colour service in 1969, this quality control system was reinforced by the introduction of Codes of Practice for the technical performance of television studio and outside broadcast installations. These Codes were drawn up in consultation with the programme companies. They have subsequently been combined into a single document containing standards for the day-to-day performance of all installations which originate or otherwise deal with programme signals. IBA Quality Control engineers make frequent visits to all studio centres to verify from technical measurements that the required standards are being maintained.

The situation with the sound radio service is rather different. The IBA has delegated responsibility for routine technical quality assessment and fault reporting to the programme companies themselves. Even with only 19 companies on the air, as at present, there are approximately 2600 hours per week of separate programme origination for which the Authority is responsible under Act of Parliament. Since the programmes are not monitored by IBA staff, it is essential for the Quality Control Section to maintain close liaison with the engineers of the programme companies. This liaison must include performance tests on the studio installations as well as consideration of day-to-day operational standards and monitoring arrangements. The work involves regular visits by IBA Quality Control teams to all the studio centres, together with working party and committee meetings, informal discussions and much correspondence. Special investigations and quality checks of broadcast

transmissions and recorded material are made as necessary.

When the television service was first established in 1955, the Independent Television Authority (ITA — later IBA) had no Quality Control Section and, although for the first few months liaison engineers were based by the Authority at the studio centres, the ITA had very little involvement in studio planning or in daily studio operations. The Code of Practice mentioned earlier was introduced about 13 years after the service had started, and considerable re-engineering was necessary to meet the new standards. Fortunately, the time-scale of the introduction of ILR enabled the IBA to draw up an appropriate Code of Practice long before applications were invited for the first contracts.

In 1972 there was uncertainty as to whether the ILR VHF services would be largely in stereo or that most studio installations would be capable of monophonic origination only. Consideration was given first to the preparation of two separate Codes of Practice, the one for stereo installations and the other for mono.

Eventually it was decided that a single Code could accommodate both sets of requirements, by the inclusion of certain additional parameters for stereo installations which could be disregarded for monophonic ones. The concept is still applicable, but it is interesting to note that not one of the first 19 companies planned a monophonic installation despite the financial uncertainties which lay ahead. This surely reflects credit on the Managements of all the companies. Stereo must have helped considerably to popularise the domestic 'hi-fi' installation and to encourage the rather stubborn British public to change to VHF reception. It must also be to some extent true that the companies reaped the justly-earned rewards which came from the greater impact of commercials in stereo. Shortly after the ILR studio Code was first published there was some adverse comment in the technical press. One writer asked sarcastically if the IBA Chairman had a 15 kHz-capable installation in her bathroom! As the years went by, less and less was said along these lines and it is a matter of some satisfaction to the IBA that the standards once thought by some to be unreasonable are now accepted as being about right.

ACOUSTICAL TESTING

(a) General Considerations
One of the matters which required special attention in compiling the original ILR Code of Practice was that of studio acoustics. The television Code has no section devoted to this subject because the television studios already existed when it was written; and, in any case, such structures are normally custom-built and architect designed, and they present very few acoustical problems apart from noise.

With ILR, it was clear from the outset that studios would be acquired by a number of different means and that many if not most of them would be converted premises, originally intended for quite different applications. Furthermore, studios intended for radio broadcasting are very much smaller than television studios, and it was necessary to consider the implications this would have in relation to reverberation times. Lastly, it was desirable for all programme companies to locate their studio complexes as close as possible to the centres of the cities and towns, and this carried the attendant problem of noisy environment, possibly incorporating a mixture of heavy road traffic, railways and underground trains, aircraft and industry. In the initial attempt to quantify the acoustical standards, the problem was broken down into the two aspects, isolation and reverberation.

(b) Isolation
The requirements for acoustic isolation depend upon the function of any particular area. To simplify the Code as far as possible, two standards have been agreed; a fairly stringent one relating to all studios and control rooms which have the capability of direct 'on-air' broadcasting, and a less stringent one relating to quality check rooms, control rooms and 'announcer' booths where the 'close-microphone' technique is used, ie, where the maximum microphone distance is 30 cm. The specification of isolation requires the establishment of maximum pressure levels which can be tolerated in the area of concern. The tolerable pressure level for a particular degree of subjective isolation is found to vary with frequency; and the curves showing this relationship are called noise-criteria or 'NC' curves. The NC20 and NC30 curves are shown in Fig. 1.
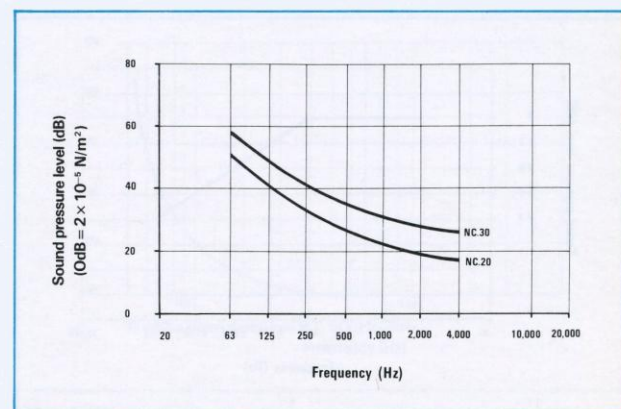

Fig. 1. Noise criteria.

The NC20 curve is the fairly stringent one referred to above, and the NC30 curve is the more relaxed standard applicable to quality check rooms, etc.

The vertical axis represents sound pressure level in dB, where 0 dB is 2×10^{-5} Newtons per square metre. It should be noted that the horizontal axis represents octave bands of random noise and not single-frequency components.

To test the isolation of a studio, it is necessary to establish that the appropriate NC curve is not exceeded with the normal lighting, ventilation plant and ancillary equipment fully operational and with normal conditions in the adjacent areas. To ensure that the adjacent areas are producing an appropriate level of sound, 'pseudo-pink' noise is generated in accordance with the spectral distribution shown in Fig. 2. (It should be noted that 'white' noise contains equal energy per unit bandwidth whereas true 'pink' noise contains equal energy per percentage bandwidth, thus falling in amplitude at 3 dB per octave as frequency increases.)

Since there are usually several different adjacent areas, it is necessary to consider not only the function of the studio under test but also the function of each adjacent area. 'Pseudo-pink' noise is then produced by means of a random noise generator, a graphic equaliser, a 250 W amplifier and four high-power loudspeakers.

The level of this noise is adjusted in the adjacent areas so that the factor 'N' in Fig. 2. is appropriate to the function of each area. For adjacent areas such as corridors, offices, shared sound-locks and talks studios, N is 70 dB. For control rooms, presentation studios and listening rooms, N becomes 80 dB. Where the adjacent area is a music studio, N is 100 dB.

It is first necessary to measure the generated noise level at several locations in each area to ensure that the level and spectral distributions are correct. This involves measurement and adjustment at about five points in each area and at eight octave bands, between 63 Hz and 8 kHz, at each point.

When the correct level of interfering noise has been set up in each adjacent area, measurements may be made in the studio under test. Again, about five sets of measurements are made, one in the centre of the studio and the others individually at a few feet from each of the corners of the room.

The average of the five readings in each octave band is taken as the result for each frequency. The normal test procedure, for each separate studio, is of first calibrating the noise meter and microphone by means of a 'pistona-phone' which provides an accurately controlled sound pressure level. Readings are then taken at each of the several points mentioned above, using octave filters in conjunction with the noise meter. These tests are made with no excitation of the adjacent areas and with the ventilation plant inoperative. The average results so obtained provide a measurement of the ambient noise level. The ventilation plant and all normal ancillary equipment is then powered, and further readings are taken to derive an assessment of the noise level under normal operating conditions, but without deliberate excitation of any adjacent areas. Finally, each adjacent area is excited as described earlier, with the plant still operating, and separate measurements are taken to assess the overall isolation.

Fig. 3 illustrates a typical ILR studio complex. It can be seen that five sets of measurements are required to assess fully the acoustical isolation of the installation since there are four studios and one control/monitoring room.

In view of the large number of measurements which need to be made, it is not normal practice to assess complete installations by routine procedure. Including the setting up of interfering noise levels, a total of about
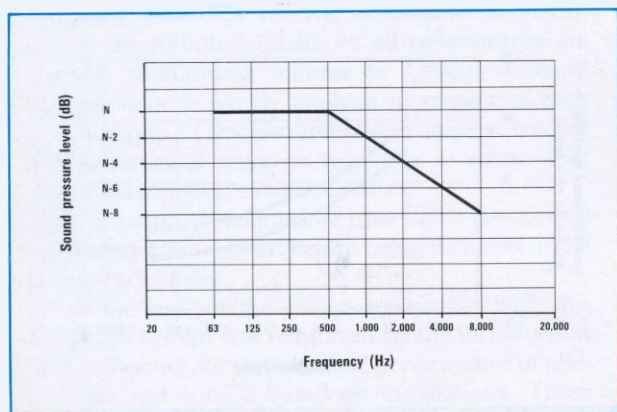

Fig. 2. Normalised sound spectrum for interfering signals.
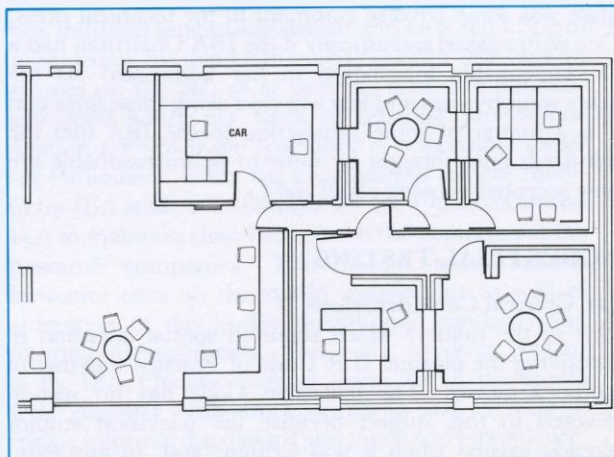

Fig. 3. Typical ILR studio complex.

1100 separate measurements would be required to test fully the isolation of a complex such as that shown in Fig. 3. It is normal procedure to examine doors and windows to ensure that they are fitted correctly and to check that door seals and closure mechanisms are working correctly. Full measurements need to be made only on new installations, new studios or booths, and in cases where inspection reveals a likely defect.

(c) Reverberation

It is necessary to consider three main aspects of reverberation which require separate specification. These are the mid-band reverberation time, the difference in reverberation time between adjacent octave bands and the maximum level at which secondary reverberation occurs. A further problem is the control of reverberation time at low frequencies in small studios, where room dimensions and sound wavelengths are of the same order.

Reverberation time is defined as the time taken for the sound pressure to decay to one thousandth of its initial value. It is proportional to the volume of the studio and inversely proportional to the total sound absorption. Reverberation time is measured by recording the decay characteristic when a broad band of random excitation is switched off. Fig. 4 shows the maximum permissible reverberation times which are related to studios and quality check rooms of a given volume, in the mid-band range between 500 Hz and 2 kHz.

This curve shows the *maximum* permissible reverberation times; but the Code contains a recommendation that figures about 10% less than these should be taken as design criteria. The Code also requires that reverberation times for adjacent octave bands within the range 250 Hz to 4 kHz should differ by not more than 10%, since greater differences would introduce audible colouration.
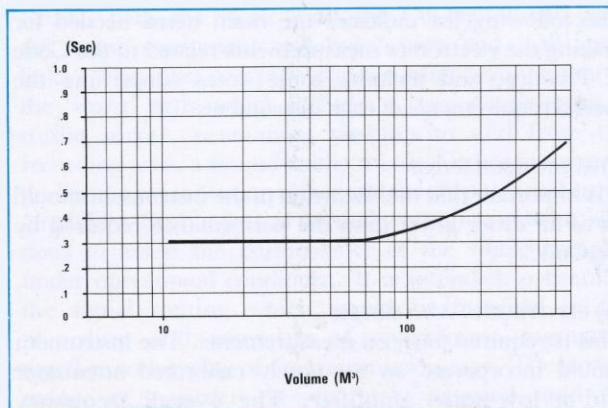
Because of the problem of reverberation times in small studios at low frequencies, the Code includes specific requirements related to studios up to 120 m³ in volume. Fig. 5 shows the maximum permissible increase in reverberation time at frequencies below 500 Hz.

The percentages shown in the vertical axis of Fig. 5 are related to the maximum reverberation time measured in the mid-band range, 500 Hz to 2 kHz. The Code requires also that the low frequency rise in studios of volume larger than 120 m³ should be reduced to the extent that no low frequency rise occurs in studios larger than 300 m³.

The sound pressure decay characteristic is of exponential form and the measuring equipment employs linear/log conversion to derive either an oscilloscope trace or a chart recording which provides a logarithmic representation of sound pressure level and yields a linear decay presentation. This sound pressure scale may be calibrated in dB. The Code requires that the decay characteristic should be of constant slope down to at least 30 dB below the initial level. Changes in the decay slope would indicate so-called 'secondary reverberation' and the existence of this phenomenon at a significant level would introduce colouration.

Quality Control Section measures reverberation by means of a chart recorder. The studio is excited in either octave or one-third octave bands over the range 63 Hz to 8 kHz. From 500 Hz to 8 kHz it is sufficiently accurate for normal purposes to use whole octave bands; but the use of one-third octaves below 500 Hz provides a more accurate analysis of the bass-rise characteristic. The level of excitation used needs to be as high as possible to enable measuring of the decay slope over an adequate range. Since reverberation time is defined in relation to a sound pressure ratio of one thousand, the corresponding curve
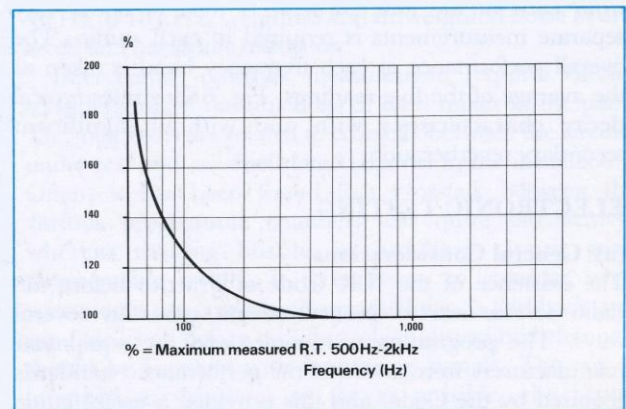

Fig. 4. Maximum permissible reverberation times, 500 Hz to 2 kHz.

% = Maximum measured R.T. 500 Hz–2 kHz

Frequency (Hz)

Fig. 5. Maximum increase in reverberation time below 500 Hz for studios up to 120 m³ volume.

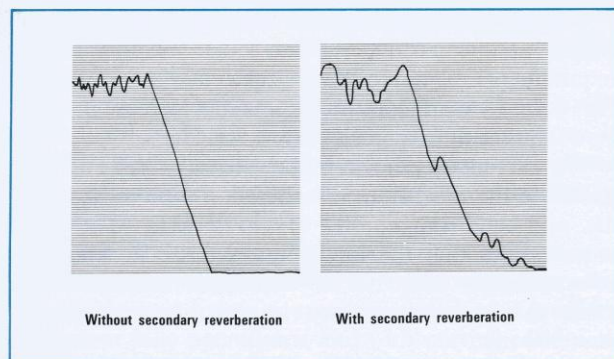Without secondary reverberation   With secondary reverberation

Fig. 6. Reverberation characteristics.

would extend over a range of 60 dB. It would be impracticable to measure over such a range since it would extend below the ambient noise level. In practice, the excitation level used is about 100 dB, or about 20 micro-Newtons per square metre, and the measurement is made over a range of 30 dB. The time reading is then doubled to obtain the result in relation to 60 dB.

The chart recorder is supplied from a microphone through an analyser incorporating an octave or third-octave filter appropriate to the excitation frequency band. The linear to logarithmic conversion takes place in the recorder itself. The recorder is started and the excitation is switched off. The decay slope is recorded on graph paper having an overall range of 50 dB; and the slope of the required section, occupying the 30 dB range, is assessed by means of a special protractor. This protractor is calibrated to allow for the reduced range and the speed of the paper, so that reverberation time can be read off directly. Inspection of the decay characteristic will reveal the presence of secondary reverberation as a change in slope. As with isolation, it is normal to measure reverberation time in the centre of the studio and at a few feet from each of the four corners. A total of about fifty separate measurements is required in each studio. The overall performance at each frequency band is taken as the average of the five readings. Fig. 6 illustrates typical decay characteristics with and without significant secondary reverberation.

ELECTRONIC TESTING

(a) General Considerations

The existence of the ILR Code of Practice before the radio service started proved advantageous in several ways. The programme companies and the equipment manufacturers were aware of the performance standards required by the Code, and this provided a useful guide when selecting equipment and planning technical installations.

As soon as the first programme companies had been appointed, regular meetings were held between their Chief Engineers and IBA engineering representatives. This liaison was formalised at a later stage, when the number of programme companies had reached double figures, and the Radio Technical Consultative Committee came into being. A Working Party, appointed by the RTCC, was given the task of reviewing the Code of Practice to ensure that it reflected the needs of the service in a satisfactory and practical way. The Code has now achieved the status of a consultative document which has benefited greatly from the knowledge and experience of the programme company engineers.

In several ways the ILR Code is a more thorough document than its television counterpart, simply because it was possible to establish standard practices at an early stage. It has been advantageous to include references to CCIR/IEC recording characteristics and peak flux levels for magnetic recording. Similarly, the opportunity was taken to explain in some detail the measurement techniques associated with the various parameters. Operational aspects also are included in the ILR Code, and among these requirements are the dynamic ranges appropriate to different types of programme material, the use of logging recorders, and studio monitoring arrangements.

(b) Test Equipment

The measurements made within a studio complex may be divided into four main categories, viz: signal paths, disc reproducers, reel-to-reel magnetic tape machines and tape cartridge machines. Certain tests such as amplitude-frequency response and signal/noise ratio are common to all four categories, while other measurements are specifically related to the equipment under test. In order to maintain a studio installation to the appropriate standards it is essential to have an adequate range of test equipment. The following list indicates the main items needed for making the electronics measurements related to the Code of Practice, and includes some notes concerning the special requirements of each instrument:

(i) AUDIO OSCILLATOR
It is important that the distortion of the instrument should be of an order lower than the performance required by the Code.

(ii) ELECTRONIC VOLTMETER
This is required for level measurements. The instrument should incorporate an accurately calibrated attenuator and a low-noise amplifier. The overall frequency response should be within ±0·1 dB over the nominal audio range, 22 Hz to 22 kHz.

(iii) NOISE MEASURING SET

This instrument should incorporate a quasi-peak meter (currently, a standard PPM to BS4297) together with an accurately calibrated attenuator, a low-noise amplifier and a CCIR weighting network.

(iv) SELECTIVE FILTER

For measurement of crosstalk and harmonic distortion in the presence of noise.

(v) DISTORTION FACTOR METER

The instrument should incorporate a variable notch filter and a meter calibrated either in dB or percentage. It should be used only when noise and hum are known to be relatively insignificant.

(vi) DOUBLE-BEAM OSCILLOSCOPE

For measurement of stereo phase difference.

(vii) WOW AND FLUTTER METER

The instrument should incorporate a weighting network to CCIR Rec.409–2.

(viii) RUMBLE METER

The instrument should incorporate a weighting network to BS4852.

In addition to this measuring equipment, it is necessary to hold a set of DIN test records for the assessing of disc reproducers.

(c) Measurement Procedures

It is worthwhile to consider a few special factors which are relevant to ILR studio testing and to review the more significant problems which have arisen during the first five years of operation.

Signal path tests fall into five categories, viz, the 'studio' path, the 'worst' path, the 'OB equipment' path, the 'news equipment and links' path and 'OB radio links'. The most complex studio configuration is that of the 'worst' path, which includes a microphone input, a studio mixer, permanent tie-lines to and from the recording area, a second studio mixer, further permanent tie-lines, a presentation mixer and the output distribution amplifiers. Such a path is normally set up at new installations to assess the performance of the whole complex under operational conditions. It is intended to simulate the signal routing which would be involved in the recording and assembly of programme segments, together with replay of the transmission tape. The performance of recorders is separately assessed, and the tolerances related to the 'worst' path are based on the combined impairments of the various mixers, tie-lines

and amplifiers. The input to the first studio mixer is at −50 dBu and −70 dBu to correspond with typical microphone levels, whereas the input to the second one is at line level (0 dBu).

Normally, a 'worst' path test can be applied only when the service has been closed down; and, when testing an operational station, it is more convenient to make 'studio' path measurements. The studio path normally consists of a single mixer, and this can be measured by using a variety of inputs at microphone level and at line level, at a number of studio outputs. Before making such tests, it is necessary to ensure that all the stereo panning controls ('pan-pots') are accurately centred, and that the channel equalisers are set to the 'flat' condition (or switched out of circuit, if suitable provision has been made). The faders must be set to their normal operational positions and, if necessary, the calibration of all PPMs must be checked and corrected. The relative levels of the stereo and mono channels should be checked at the output to ensure that the mono level is 3 dB above that of the A and B channels. The output levels obtained from the amplitude-frequency test must be corrected to allow for any variation of input level.

The 'studio' path and 'worst' path tests include gain adjustment, gain stability throughout one hour, frequency response from 40 Hz to 15 kHz, weighted and unweighted noise at three input levels, interchannel crosstalk, harmonic distortion, A/B level difference, A/B crosstalk, A/B phase difference, output impedance at 50 Hz, 1 kHz and 10 kHz, and output balance. Similar parameters are measured on the OB equipment path; but, in this case, the input balance also is measured. The news equipment and links path is tested at one level over a restricted frequency range from 300 Hz to 5 kHz, and the gain stability, interchannel crosstalk and stereo tests are not required. The OB radio links tests include gain adjustment, gain stability, frequency response from 40 Hz to 10 kHz, weighted and unweighted noise at one level, and harmonic distortion.

Interchannel crosstalk measurement requires careful consideration of the functions of each mixer in the path. The tolerances are related to crosstalk between the circuits under test and *any* dissociated channel in the installation. Often, it has been found that crosstalk between the various programme channels was quite satisfactory, whereas making hostile an auxiliary circuit gave unacceptable results. It is necessary to consider what operational use is made of such facilities as fold-back and pre-fade listen before deciding which dissociated channels should be supplied with interfering signals. The test is intended to simulate, so far as possible, the actual conditions which prevail during programme operations.

One of the harmonic distortion tests is to ensure that

adequate 'headroom' exists in the programme channel amplifiers of studio mixers. The first tests are made with an input line-up level of −70 dBu, with normal attenuator and channel, group and main fader settings. The input level is then increased to −50 dBu, and the output level is restored to 0 dBu by means of the channel fader only. This technique ensures that the stages preceding the channel fader are subjected to a level 12 dB above the normal maximum, which is a situation likely to be encountered occasionally in practice.

The original ILR studio Code set common standards for both broadcast cartridge machines and reel-to-reel tape machines. At first sight, this approach seemed reasonable because the tape speed and dimensions are the same in both cases. In the light of experience, it has been necessary to introduce more relaxed tolerances for cartridge machines. Furthermore, cartridge machines are precluded from the full range of programme applications unless their performance is within the reel-to-reel limits. The problems with cartridge machines are largely due to the tortuous tape-path and the associated mechanical difficulties. Wow and flutter tend to increase due to variation of 'stiction'. Frequency response and stereo phase stability are impaired by tape weave. The machines need frequent and careful maintenance and alignment, particularly with regard to azimuth adjustment, cleaning and replacement of the capstan, the pinch rollers and the heads. The performance of the machines is worse with cartridges of longer duration; and, for this reason, it is fortunate that their main application is the origination of commercials, promotions and jingles. Some improvement has resulted from the use of lubricated tapes; but, with those versions currently available, the distortion noise/performance is unsatisfactory. It seems that the NAB 'A' format is not really good enough for full broadcast applications, and only a radically different design would yield significant improvement.

The professional reel-to-reel machines used in ILR present very few problems, provided that they are well maintained. Nevertheless, there is still room for improvement in high frequency phase stability and for a reduction of tape hiss.

Disc reproducers currently available are generally more satisfactory than were earlier ones. Rumble has been a problem with the less expensive machines, and all disc reproducers are sensitive to structure-borne vibration. Only a heavy and stable support will eliminate this difficulty. Excessive wow and flutter were very common faults in the early days of the ILR service, but the introduction of reasonably-priced machines of the direct-drive type has largely resolved this problem. One remaining impairment, which is encountered occasionally, is excessive crosstalk between the stereo A and B channels. This problem is invariably due to the pick-up cartridge being either faulty or of poor design.

(d) Operational Checks

Since the companies are required to monitor their own programmes, Quality Control Section must ensure that adequate facilities exist at each studio centre. The programme companies have been asked to install professional MF and VHF receivers, with adequate aerial installations, since the main monitoring depends on off-air signals. As yet, specific performance limits for monitoring receivers have not been established. This is because of the variation between sites, the uncertainties of propagation conditions at different times and seasons, and the hazards of co-channel and adjacent channel interference. Instead, Quality Control Section conducts so-called 'loop' tests, in which signals are originated at the studio centre and received off-air at the VHF transmitter site and at the studio. This procedure confirms that the high-quality VHF service is adequately received at the studio centres, where it forms the primary monitoring source. Further checks are applied to verify that suitably placed loudspeakers, of approved type, are available in all areas where monitoring is performed, and that the MF service also can be satisfactorily received.

Checks are made to ensure that programme volume is controlled in accordance with the requirements of the Code. This aspect of ILR operations has been especially troublesome in the past, mainly because most programme presenters are non-technical people and need to be persuaded to give proper attention to the dynamic range of the studio output. Thanks mainly to the efforts of the programme company engineers, the situation has now improved; and it appears to be getting still better as the staff gain experience. Recent modifications to the Code of Practice require the installation of audible or visual indication of over-modulation at points of volume control. In this connection, the advent of peak programme meters using columns of light-emitting diodes seems likely to be of relevance. Instruments are now available with LEDs emitting green light for levels of and below +8 dBu and red light for higher levels, thus providing clear and unambiguous indications.

A further operational matter of concern to Quality Control Section is the use of slow-speed logging recorders. A continuous recording of the output of the VHF and MF transmissions is required by the Authority for numerous non-engineering purposes; and programme companies must install the necessary equipment. Experience has shown that, unless precautions are taken, the technical quality of such recordings can easily deteriorate to the level where speech becomes unintelligible. The latest version of the ILR Code

includes a basic day-to-day technical performance specification for the logging machines, and this aspect will from time to time be checked by Quality Control engineers. it is hoped that the new procedure will lead to the necessary improvement in the recordings.

FUTURE PROSPECTS

(a) The Need for Changes
Quality Control Section is always seeking better and, if possible, quicker ways of performance testing. The number of stations is expected to increase, throughout the next five years or so, at an average rate of approximately one every six weeks. Unless new techniques are introduced, the Section will be forced to reduce the frequency of its visits to each studio centre or, by 1984, to quadruple the number of engineers involved. It is necessary, therefore, to consider two aspects of future development. First, there is the possibility of introducing new techniques which will improve the accuracy of assessment. Secondly, there is a very strong incentive to seek ways of accelerating test procedures, provided that accuracy is not sacrificed. Ideally, the new methods would be more accurate besides being quicker.

(b) New Methods of Measuring

(i) AUDIO NON-LINEARITY
One of the more time-consuming tests is in the measuring of harmonic distortion. Tests are made on signal paths, disc reproducers and tape recorders; and, in some circumstances, numerous different measurements are required to cover the full range of investigation. Despite this effort, there is poor correlation between the measurement and the subjective impairment. It is said that an experienced listener can detect 1% second harmonic distortion at 1 kHz but is equally aware of 0·4% third harmonic. On more complex signals, the same listener would detect total harmonic distortion equivalent to 0·2%.

Quality Control Section are considering some new proposals for the assessment of audio non-linearity which would replace measurement of harmonic distortion.

(ii) MODULATION NOISE
Another type of measurement currently under consideration concerns an impairment known as 'scrape flutter'. This is actually a form of modulation noise which occurs in magnetic tape recording. The mechanisms which give rise to it are complex, but the impairment manifests itself in the form of groups of spurious sidebands closely spaced around the wanted fundamentals. Several years ago, ATV ran multigeneration tests before placing a large

order for $\frac{1}{4}$-in tapes. After about five generations they encountered both objective and subjective distortion. It was significant that the recordings giving the worst subjective quality were not those giving the worst results on standard tests; and the main defect was of harshness due to modulation noise. Quality Control Section are examining this problem in the hope that more will be learned about the machine/tape combination and that a measurement technique and performance objectives can be established.

(c) Computer Controlled Testing
During any typical Code of Practice visit, much more time is spent in travelling, setting up equipment, writing down test results, normalising measurements and producing a typed report than is devoted to testing. Furthermore, the time occupied in measuring is largely spent on the mechanics of the process rather than on objective assessment of equipment performance.

Recent developments in micro-processing technology suggest that audio testing could be performed less arduously. Conceivably, a computer could be used for applying and controlling the measurements, and also for producing the printed test results. This would allow the engineers more time for constructive thought. It would also increase their productivity and enable the tackling of a larger volume of useful work. Automatic measurements could be made at sufficient speed to be evaluated on the spot, and repeat tests could be undertaken at will.

A certain amount of equipment for this purpose is already on order, and there is likelihood that the first phase of computer controlled testing will soon be introduced. The control unit is a proprietary micro-computer which uses floating point BASIC language. The test program and data storage will be on a floppy disc, and the output will be available on a visual display unit. Probably the data will be transferred, at Crawley Court, to a mainframe computer which would provide a hard copy output on the line printer together with long-term storage of results and analyses.

The proposed measuring system will incorporate the following programmable elements:

(i) Audio oscillator
(ii) Switch unit
(iii) Level and noise meter
(iv) Frequency counter
(v) Tuneable filter
(vi) Phase meter
(vii) Distortion Measuring Set.

An RC oscillator was chosen because of the possibility that a synthesiser might introduce distortion. The design includes reed relay switching of passive com-

ponents to achieve frequency and voltage control. Frequency selection will make use of an eight-bit binary code, giving 256-step resolution, but a ten-bit code for level control will be necessary for obtaining 0·1 dB resolution from −70 dBu to +8 dBu.

The switch unit will provide for selection of programme, oscillator or termination, to the input of the system under test. It will also select, for input to the measuring equipment, the input source, the stereo A output, the stereo B output or the mono output. Level and noise measurements will make use of an existing test set. Phase measurement will be made by means of a new phase meter which has a very rapid response, thus enabling statistical analysis of a large number of tests to enable better understanding of phase flutter on tape machines.

The method of distortion measurement is still under consideration; but, where noise is insignificant, a normal distortion factor meter will be used. For low-level paths, and for possible intermodulation and modulation noise tests, it is the intention to use a spectrum analyser. Since this is a manual instrument, the test program will pause to allow for setting of the front panel controls. On command, the computer will read-off the results and present them on an X–Y plotter, thus providing greater accuracy than the existing method which involves measurement on a small CRT.

The current proposals will provide very valuable experience in return for a modest investment. The detailed costs of a complete system are not yet known. Nevertheless, if the productivity of the Section can be increased to the extent that seems possible, the saving in revenue expenses will rapidly overtake the initial capital outlay.

A Contribution Network for ILR

by C W B Reis, R Francis and B F Davies
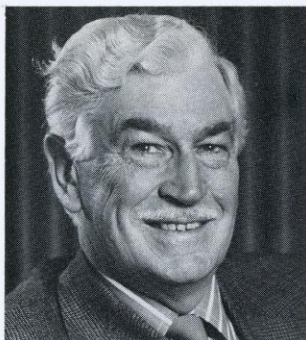
WILLIAM REIS, Bsc. graduated from London University in 1951 and was engaged in the development of television transmitters and studio equipment at EMI Research Laboratories Ltd. He was then employed by Rediffusion Television Ltd., becoming Chief Engineer, Planning. He joined the Authority in 1969 where until recently he was technical editor of *IBA Technical Review*. His present position is Head of Lines Section.

ROGER FRANCIS attended college in Surrey from whence he joined the BBC External Services as a Control Room Engineer at Bush House. With the start of Independent Local Radio in 1973 he joined London Broadcasting Company/ Independent Radio News and was appointed Head of Engineering in June 1976. Being a member of three local radio committees, he is much involved with the technical developments of ILR and has been instrumental in the planning of the IRN contribution network. He is married with two children and lives in Kent.

BRIAN DAVIES, MIEE, was engaged on the development of TV and audio wired distribution systems prior to his joining The Marconi Company in 1950 to work on marine aerial distribution systems. In 1955 he joined Associated-Rediffusion to head the links and communications section and later was project leader on studio planning. He held the post of Chief Engineer with Grampian Television for four years from 1961, and after a period as Assistant Chief Engineer with Rediffusion (SW) at Bristol he joined the Authority. As Senior Lines Engineer he has since been concerned with the operations and planning of the ITV and ILR networks.

Synopsis

National and international news is constantly available to all ILR companies from Independent Radio News (IRN) which operates in conjunction with London Broadcasting Company (LBC) in London. Regular news bulletins are sent out from IRN by means of a distribution network of Post Office circuits radiating from London to all ILR companies. This network has been in operation for some time past and will be extended as additional ILR stations come into service.

An additional requirement has arisen whereby the ILR companies can contribute news of local events for re-distribution to some or all of the other ILR stations. For this purpose a new network, to be known as the contribution network, is currently being established for linking each ILR company direct with IRN/LBC in London. To economise on Post Office circuits it will consist initially of seven spurs or branches arranged in such a way that the more remote companies on each spur will be linked via those nearer to London. Therefore, at all companies other than those at the extreme ends of a spur there will be a need for switching to enable individual circuits to be connected in tandem as required.

The switching system which has been designed makes use of self-routing tones which, when injected into the network at any studio requiring to contribute a news item, will automatically activate all switches between that studio and IRN/LBC, thus providing a continuous path. The routing tones used for this purpose are under the control of IRN/LBC and are distributed to the various ILR studios via the national STD telephone network.

INTRODUCTION

The overall configuration of the Independent Local Radio (ILR) network falls neatly into three discrete categories. The first, which might be termed the 'trans- mitter network', consists of all the separate connections that link the studios of the different companies to their appropriate transmitting stations. It should be noted that, within the service area allocated to it, each ILR
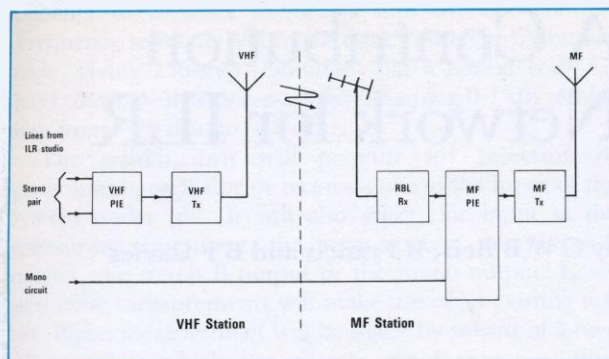
Fig. 1. The signal paths from the studios of an ILR programme company to the programme input equipment (PIE) of the IBA VHF and MF transmitting stations, usually in different geographical locations, are provided by Post Office circuits as shown. Should there be a failure of either the A or B stereo component the VHF service will continue in mono. A re-broadcast link is available against possible breakdown of the circuit feeding the MF service.

programme company provides a duplicated service in two bands — a high quality stereo service in VHF, and a bandwidth-limited mono service in MF. Hence, there is normally a single mono line to an MF transmitting station, and a pair of stereo lines to a VHF transmitting station at which is located the associated stereo encoder (Fig. 1). The use of two separate circuits for carrying the A and B components of the stereo signal avoids the difficulty and cost of equalising a single circuit for carrying encoded stereo. Further, it affords a measure of protection in that the service can be maintained in mono should there be a failure of either circuit. In the MF case, a reserve programme feed is available from a dedicated receiver tuned to the associated VHF transmitter, thus forming a re-broadcast link (RBL).

However, the gathering of national and international news is an expensive and highly specialised activity, and it would be very uneconomical if each programme company separately were to gather its own news. For this reason it was made an essential part of the London Broadcasting Company (LBC) contract that it should gather the news on behalf of ILR as a collective body. Therefore, a means had to be provided for disseminating news material from a part of LBC, known as Independent Radio News (IRN), in London to all the other ILR companies, and this takes the form of a system of Post Office circuits radiating from IRN. It is known as the 'distribution network' and constitutes the second category of the complete network. The distribution network has been built up, and indeed will continue to expand, with the growth of the ILR service. There is likelihood that, by the end of 1981, it will embrace 34 ILR programme companies.

NEWS GATHERING

A little more than a year ago the proposal was made to establish a second system of PO circuits similar in configuration to the distribution network but operating in the opposite direction. In other words, instead of it being a network diverging from London it can be thought of as one converging towards London. Its purpose is to enable individual companies to contribute to IRN information concerning events occurring locally but having more than merely local interest. Such information can then be used immediately by IRN for inclusion in the news output of the LBC from the London transmitters. Also, it can be passed on to any or all of the other companies via the distribution network. This constitutes the third category of the network and is referred to logically as the 'contribution network'. It is still very much in the course of being established; and, as with the distribution network, will continue to grow in accordance with the needs of ILR as a whole.

BASIC PRINCIPLES

The most straightforward way of fulfilling this need is to provide a separate line from each company to IRN, but as these lines are required only for use singly rather than collectively this method would be somewhat extravagant. A scheme has therefore been devised whereby all companies excluding Capital Radio will be connected to one or more (maximum 12) trunk circuit routes or spurs into IRN. Capital Radio, the company which provides general entertainment programmes in the London area, is excluded for the obvious reason that news of events in and around London is covered directly by IRN/LBC. Figs. 2 and 3 show the contribution network as it is expected to appear by the end of 1981 and early 1983 respectively, and Table 1 shows the list of abbreviations used for the locations indicated. From these it will be seen that, for contributions to be received in London from some of the more remote companies, a switching operation is necessary at one or more points along the route. A system for doing this, and believed to be entirely novel, is currently under development. It will comprise special apparatus that will need to be installed in the premises of each programme contractor and which will operate in conjunction with a dedicated exchange line (DEL) and a telephone available at the news desk. In each case, the telephone instrument will be fitted with an automatic call-maker whereby calls can be made to IRN only. Apparatus will be needed also at IRN, but of a type essentially different from the rest; and, in this case, there will be eight dedicated exchange lines available at three positions, each provided with a key-and-lamp unit, a telephone instrument and initially a Post Office 'XL' call-
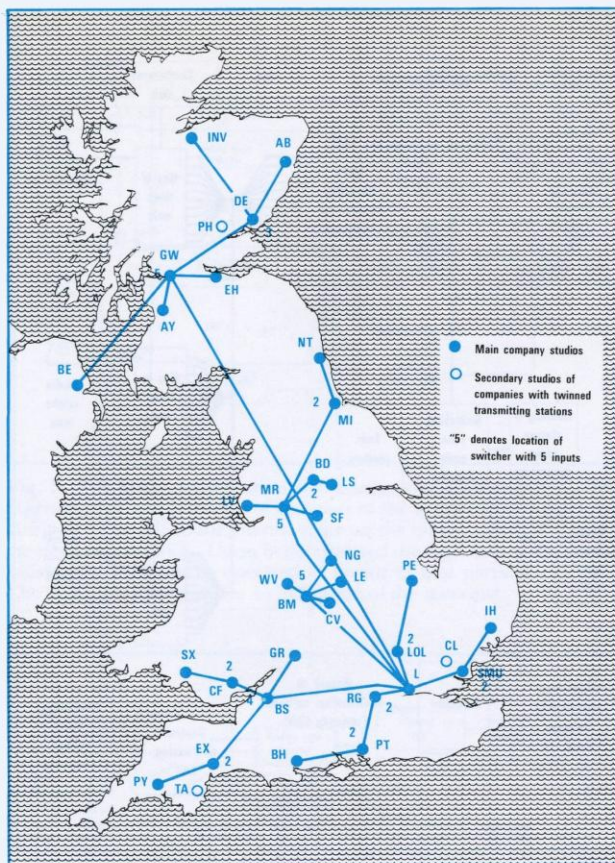
Fig. 2. A contribution network enabling ILR companies to pass news information to Independent Radio News (IRN) in London is being established and will grow as new companies enter service. The diagram shows the extent of the network as it is expected to be by the end of 1981 embracing 34 companies.

The news editor informs all companies, by teleprinter, of the cueing and other relevant details; and, at the next scheduled time for a news feed (usually on the hour), the material is fed to the appropriate outgoing networks following a 30 sec burst of 1 kHz tone which automatically starts the tape recorders in the studios of the various companies.

Each receiving company then dubs the material from tape to a cartridge, collects the appropriate cue information from its own teleprinter, and the news item is ready for transmission.

A distribution network used for the disseminating of news material from IRN to the regional companies has a similar configuration. Table 1 gives the key to the abbreviations used for the locations indicated.

maker capable of storing 46 dialling codes. From any one of these three instruments it will be possible for STD calls to be made exclusively to any of the corresponding telephone instruments at the other companies. The eight IRN exchange lines will be numbered consecutively to allow automatic 'group hunting'. In this way a (Post Office) mono call-maker is all that is required at the premises of each company.
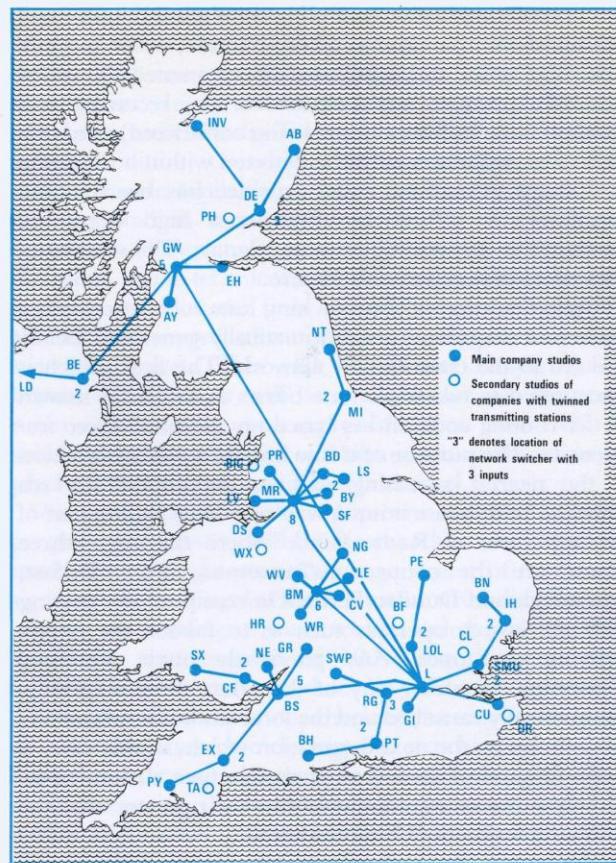
Fig. 3. By 1983 the contribution network will have been extended to provide for companies operating in 44 areas as shown here.

METHOD OF OPERATION

The switching of the contribution network will be controlled exclusively from IRN. The system makes use of self-routing tones or tone codes; and, once these have been initiated at IRN, the switching action at all points between the contributing programme company and IRN is entirely automatic.

The *modus operandi* can probably best be explained by considering an example. Suppose that an event taking place in Glasgow is thought to be of general interest, and that the local company, Radio Clyde, wishes to offer it as a news item for general dissemination to the other programme companies. Using the dedicated telephone, details are exchanged with the news room at IRN and, if the item is accepted, the switching process is initiated simply by operating a key pad associated with the IRN source selection system to be described later. By so doing, a signal comprising a duplicated triple-tone switching code is applied to the public subscriber trunk network

25

(PSTN). Individually, each tone code consists of a short burst of three tones between 1 and 2 kHz, occurring in a particular order and occupying approximately 50 ms in total. To produce any response at the receiving end (Radio Clyde in the example being considered) two such triple tone sequences must be detected within a time gate of 250 ms. This tone code sequence has been chosen specifically to afford the detectors a high degree of immunity to any speech or music signals. The security of the system depends on this feature.

Upon detection of the switching tone code, two further triple-tone sequences are automatically generated locally and fed to the contribution network. The first, which is transmitted as two triple-tone bursts as before, is known as the routing code and is based on the same three frequencies. The purpose of it is to initiate a switching action at the nearest switching unit in the circuit towards London. In this case it is a 4-way switcher forming part of the apparatus at Radio Clyde where the other three inputs are the connections incoming from Belfast, Edinburgh and Dundee/Perth. On receipt of the routing code the switch operates such as to favour the circuit carrying the tones. Any previously made switching connection favouring any of the other three sources is automatically cancelled and the local studio connected via the switcher to the next destination which, in this case, is IRN. On arrival there it is used to initiate a momentary audible buzz as a confirmation to the operator that the switching action has been completed.

The second tone code generated at Radio Clyde is the station identification code which is unique to that station. This tone code is transmitted as a single triple-tone sequence only. It includes one frequency that is different from any of the three previously used, though still within the 1 to 2 kHz band. It is produced about one second after the routing code, thereby allowing sufficient time for successive switching actions to have completed the path to IRN where its detection activates an appropriate Light Emitting Diode (LED) indicator identifying Radio Clyde as the company making the contribution. Thus, the traffic can then be passed via the network from Radio Clyde, for IRN to use as required, and the circuit will remain switched through from Radio Clyde until such time as a different switching pattern is initiated by IRN.

To take a somewhat more complicated example, the case of Metro Radio in Newcastle-upon-Tyne will now be considered. This, it will be noticed (Figs. 2 and 3), is at the extreme end of a spur. After having made contact with IRN by using the dedicated telephone, and after the switching tone code has been sent via the PSTN as in the first example, the locally generated routing tone code from Metro Radio passes via the network from Newcastle to the nearest switching unit in the direction of London.
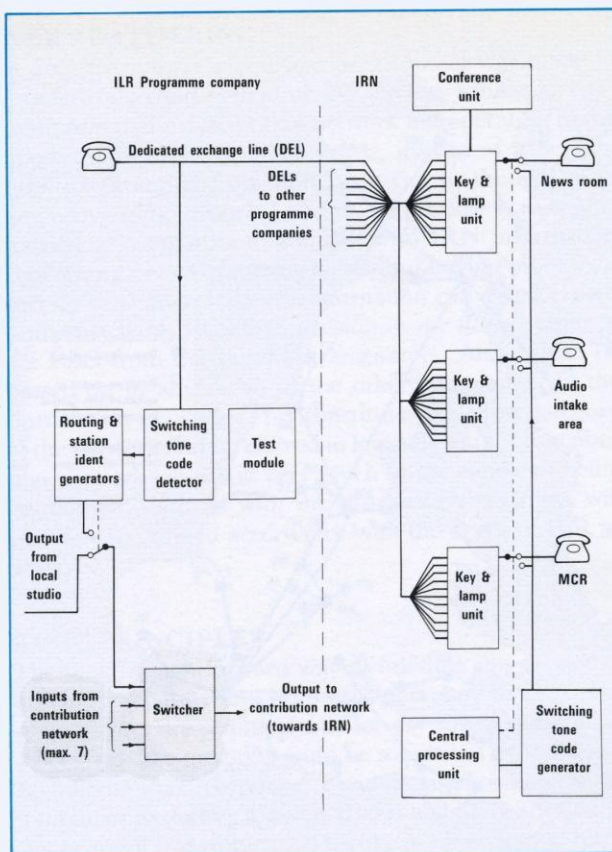

Fig. 4. Diagram of the overall control facilities for the contribution network switching system. The blocks, described in the text, are the principal items of equipment that will be required at IRN and at the studios of each programme company. But it should be noted that companies at the periphery of the network will not require the switcher and its associated circuit.

This is to be found at Radio Tees, in Stockton-on-Tees; and, on detection of the switching signal from Newcastle, it operates in favour of Metro Radio. It also applies a re-generated routing tone code to the next link of the network which terminates as an input of an 8-way switcher located at Piccadilly Radio, Manchester. As before, this switcher operates to favour the route from Metro Radio and the routing tone code is once again regenerated. This is applied to the next and final link of the network and, on arrival at IRN in London, its detection confirms by means of a momentary audible buzz that the switching action is complete. One second later the station identification tone code from Metro Radio, having passed through the now completed network, is also received at IRN and is used to activate an LED indicator providing continuous confirmation that the circuit is complete from Metro Radio. Block diagrams
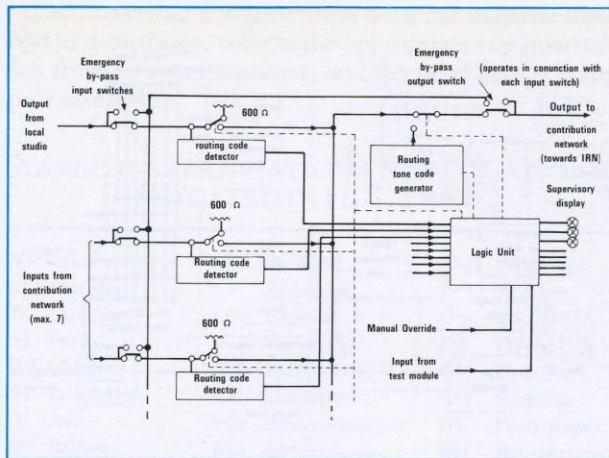
Fig. 5. The basic configuration of the tone-controlled switcher. This diagram is an amplification of that part of the system shown coloured in Fig. 4. The self-routing tones enter on the circuit which will carry the news contribution. Upon being detected the logic unit will respond causing the relays to be operated in favour of that particular circuit. The logic unit also provides an indication of the *status quo*.
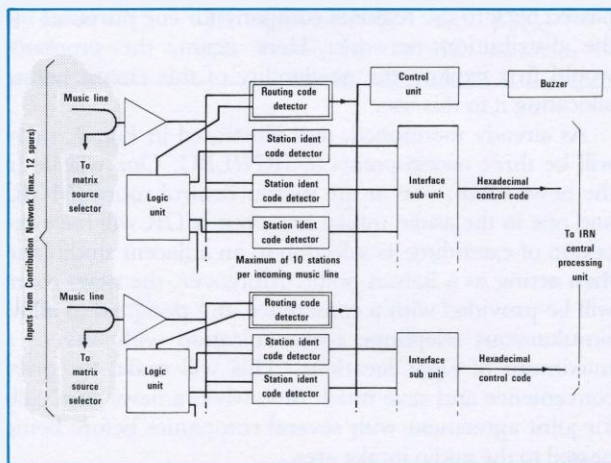

Fig. 6. Each spur of the contribution network terminates at IRN and has associated with it a routing code detector, which provides a buzzer indication that the circuit is complete. It has also a maximum of 10 station identification code detectors. The outputs from these latter are converted into hexadecimal logic before being passed on to the central processing unit.

illustrating the arrangement of the equipment and its method of operation are shown in Figs. 4, 5 and 6.

It will be appreciated that the system will allow IRN to receive more than one contribution at any time provided that the contributing companies are on different spurs of the network incoming to London. Such is the case with Radio Clyde and Metro Radio, and therefore the two examples quoted above could occur simultaneously.

EMERGENCY SWITCHING AND TESTING

Mention should also be made of the arrangements for dealing with emergency switching in the event of equipment failure. In the case of the last example, should the switcher at either Radio Tees or Piccadilly Radio develop a fault condition, it would normally be possible, provided that staff were available, to patch the lines at a jackfield and so establish manually the required network configuration. However, for convenience and to cover those occasions when technical staff are not available, it has been decided that the lines in and out of the switchers should be routed through an emergency by-pass switching panel. This will incorporate simple push-button operation. It will be available for use by news room staff, but only upon express direction from IRN. A suitable key-operated lock will be incorporated such as to preclude access by unauthorised personnel.

An integral part of the equipment supplied to each programme company will be a test module which may be used either as a detector or as a generator of tone code signals; but, once again, there must be strict discipline to prevent the sending of tones to line unless with the full knowledge and consent of IRN. To this end, the ON/OFF switch on this module also will be key-operated. The module can be used for checking the tone codes from all incoming sources including the switching tone code from the dedicated exchange line. It will also provide a means of testing the local tone code generator and will enable application of tone codes to the outgoing music circuit. With the aid of a similar test module at IRN it will be possible to isolate any fault on the contribution network.

Physically, it is expected that the apparatus of each company will be accommodated in a single 19-in frame occupying 7-in of rack height, but space must be provided also for associated jackfield facilities.

THE WORKING ENVIRONMENT AT IRN

To engineering and editorial staffs at IRN, the new system will be of great advantage. At present, incoming contributions from provincial companies are received either by telephone or via an occasional programme (OP) circuit specially rented from the Post Office. An STD telephone call can usually be established quickly, but the quality of transmission does not meet broadcast standards. Occasional programme circuits do meet broadcast standards but are not always available in the time or at the locations required.

All news reports incoming to IRN from the contribution network or from other sources in London are routed to the 'audio intake' area where they are recorded and prepared for later distribution on the network. However, with the expanding scale of the IRN news service there is

a need in the audio intake area to cater for up to 100 news sources. At present, in addition to the studios of LBC and Capital Radio, there are 24 outside studios or reporter sites in the London area and a further 12 are being planned. With the introduction of the contribution network, and with the new ILR programme companies now being planned, the number of regular news sources is likely to increase to about 80 in the next two or three years. Hence, it has been necessary also to provide a central processing unit within IRN to avoid the possibility of congestion. This unit has been developed by engineers at IRN/LBC and incorporates a source selection system based on a microprocessor. It will control the selection of the various sources of news and will route the signals into IRN.

Basically, each potential news source is allocated a code comprising a letter followed by two numerals. By means of a key pad combined with a display panel the studio engineer is able to use the code for selecting a desired source and allocating it to a spare channel on the studio mixer. Provided that the source has not already been selected for allocation to another technical area within IRN, confirmation of its availability will be indicated by the illuminating of a lamp above the appropriate channel fader and by a read-out of the code appearing on the display panel. Moreover, where appropriate, the selection of a source will make available automatically the associated cue, programme, talkback, override and other facilities which the operator may require. But, should the desired source already be assigned to another studio, a suitable indication will be made available to the operator who then will merely be given access to a 'listen' feed until such time as the source has been de-selected at the other studio.

This procedure applies equally when the desired source is an ILR company. Included among the inputs to the switching matrix are the incoming spurs of the contribution network. On being offered a news item from a provincial ILR company, the studio engineer needs only enter the corresponding code on the key pad. Provided the appropriate spur is available, the microprocessor will automatically apply the triple-tone switching code to the PSTN. The switching action then takes place as already described. If the station identification code is received correctly at IRN, the processor will allow registration of the appropriate indications. A block schematic of the IRN audio switching and control system is shown in Fig. 7.

In most normal cases the dedicated exchange line will suffice as a control circuit allowing inter-communication between the respective news rooms while the news item is being taken; but, should a two-way interview be required between, say, a reporter at the remote end and an IRN
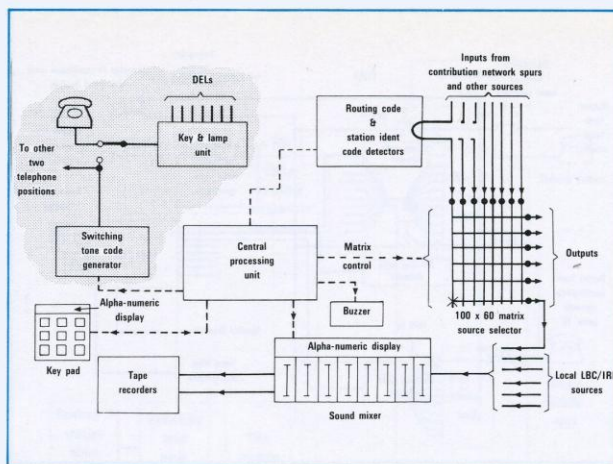

Fig. 7. Schematic diagram of IRN source selection and switching control system. The components of the co-ordination network are shown coloured. The routes of the programme circuits are shown in solid line and the dotted lines represent control circuits.

interviewer, an output from the local studio would be passed back to the regional company for cue purposes via the distribution network. Here again, the processor would first explore the availability of this circuit before allocating it to this use.

As already mentioned, and illustrated in Fig. 4, there will be three access points at IRN/LBC. One will be in the news room, one in the master control room (MCR) and one in the audio intake area, but MCR will have the option of extending its selection to an adjacent studio and then acting as a liaison point. Moreover, the news room will be provided with a conference unit designed to allow simultaneous telephone communication with any of a maximum of eight locations. This will make for great convenience and save much time when a news item calls for joint agreement with several companies before being passed to the audio intake area.

DISTRIBUTING THE NEWS

On completion of a contributed news item the STD call is terminated and the contribution circuit is de-selected by operation of the key pad thereby releasing the network spur. The news item is then edited, dubbed on to a cartridge, given a title, reference number and suitable cues.

The news editor informs all companies, by teleprinter, of the cueing and other relevant details; and, at the next scheduled time for a news feed (usually on the hour), the material is fed to the appropriate outgoing networks following a 30 sec burst of 1 kHz tone which automatically starts the tape recorders in the studios of the various companies.

Each receiving company then dubs the material from tape to a cartridge, collects the appropriate cue information from its own teleprinter, and the news item is ready for transmission.

TABLE 1: ABBREVIATIONS FOR LOCATIONS INDICATED IN FIGS. 2 AND 3

INV	Inverness	SF	Sheffield	SMU	Southend
AB	Aberdeen	LV	Liverpool	L	London
DE	Dundee	DS	Deeside	CU	Canterbury
PH	Perth	WX	Wrexham	DR	Dover
GW	Glasgow	NG	Nottingham	GI	Guildford
EH	Edinburgh	LE	Leicester	RG	Reading
AY	Ayr	WV	Wolverhampton	PT	Portsmouth
BE	Belfast	BM	Birmingham	BH	Bournemouth
LD	Londonderry	CV	Coventry	SWP	Swindon
NT	Newcastle	WR	Worcester	BS	Bristol
MI	Middlesbrough	HR	Hereford	GR	Gloucester
MR	Manchester	PE	Peterborough	NE	Newport
PR	Preston	BF	Bedford	CF	Cardiff
BIC	Blackpool	LOL	Luton	SX	Swansea
BD	Bradford	BN	Bury St Edmunds	EX	Exeter
LS	Leeds	IH	Ipswich	TA	Torbay
BY	Barnsley	CL	Chelmsford	PY	Plymouth

DEREK CHAMBERS, MIEEE, AMIERE, joined the Independent Broadcasting Authority in 1970 from the BBC, where he had been engaged on a number of projects. Among these was the design and construction of UHF relay stations and VHF radio stations in the Transmitter Capital Projects Department. Prior to that, he spent several years in industry, where he worked on the research and development of a wide range of electronic projects. His early work with the IBA was as a Senior Engineer working on UHF transmitting aerial systems, and in 1972 he became Head of the Authority's new Local Radio Project Section. Since 1978 he has been Head of the Transmitter Project Section in Station Design and Construction Department. He is married with two children and lives in Surrey.
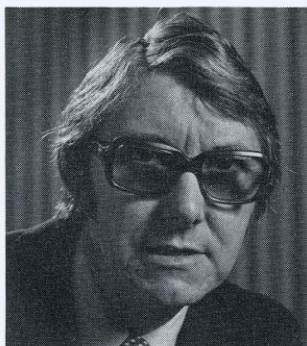
Phase II ILR Transmitting Stations

by D S Chambers

Synopsis

The author reviews the major technical features of the first Phase ILR transmitting stations and discusses some of the new concepts and equipment introduced into the coming Phase II of construction. The new solid-state MF and VHF transmitters are discussed. These provide transmitter powers of maximum 1 kW and 300 W respectively, and particular attention is given to the modulation process used by the MF transmitter. The introduction of the 'twinned' station concept is discussed in terms of the measures for overcoming problems of programme distribution and monitoring.

Circular polarisation will continue to be used for the VHF transmitting aerials and, because of the particular interest shown in this area, a review of circular polarisation principles has been included.

REVIEW OF THE DESIGN FEATURES OF PRESENT ILR TRANSMITTING STATIONS

Phase I ILR transmitting stations were designed and equipped in the period 1972–1976. Before discussing the design of future ILR transmitters it is worthwhile summarising the major technical features of the earlier Phase I transmitters and these are set out below:

(a) Transmitters

Passive reserve configurations with full power standby were used at all MF and VHF stations. In the majority of cases thermionic amplifiers were used with the exception of four of the VHF stations where all solid-state equipment was provided.

(b) Telemetry and PIE

An in-band over-air signalling system was used to transmit the status of the transmitter sites to the studio of the programme contractor responsible for monitoring the stations, both in terms of alarms and quality, and for informing the appropriate IBA maintenance control centre of any fault occurrence. Main and reserve PIE chains were provided at all stations. Two particular features were that the MF stations were provided with 5 kHz low-pass audio filters and all VHF stations were stereo capable.

(c) Aerials

VHF stations used both omni-directional and directional circularly polarised aerials. At a number of MF stations, highly directional aerials were provided to enable the re-use of the 1151 kHz and 1546 kHz channels throughout the UK. In addition, omni-directional mast radiators, approximately 60 degrees in electrical height, were provided at omni-directional stations.

(d) Power

Standby generators have been, or are about to be, provided at all MF transmitters.

(e) Equipment Housings

All the technical equipment was housed in conventional brick buildings.

(f) Reliability of Service

It is interesting to note that, despite the fact that MF and VHF transmitters were of a relatively old design incorporating thermionic devices, the reliability of the stations has proved to be extremely good. This can be illustrated by the average figures of service breakdown: for example, at MF 1 kW and VHF 1 kW stations the average breakdown times for any four-week period are currently less than two minutes.

The design of future ILR transmitting stations will retain some of the concepts of the previous Phase I transmitters. However, due to the development of new all solid-state transmitter equipment and a number of other developments, there are some significant changes in the design of these stations, and it is the purpose of this paper to describe and discuss these changes.

NEW MF TRANSMITTER — BASIC CONCEPTS AND DESIGN

Introduction

The new MF transmitters (to be supplied by Redifon) are of all solid-state design and use a broad-band (520 to 1610 kHz) modular approach in their construction. The transmitters give output powers of 1000 W and 500 W, and a pair of 1000 W transmitters can be combined to give a 2 kW output capability. The output power is achieved by the c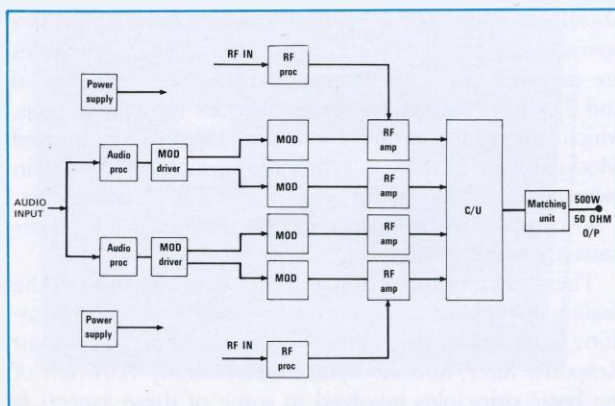ombination of a number of modules in parallel. A basic block schematic and the physical arrangement of the transmitter are shown in Figs. 1, 2 and 3. As can be seen, the transmitter uses an 'active reserve' configuration.

The transmitter uses a number of concepts which are recent innovations in their application to medium-wave transmitters which are not apparent from the basic schematic, namely Class 'D' audio amplification, which involves the concept of pulse width modulation and RF transistors operating in a switching mode. In order, therefore, to have a better appreciation of the transmitter, it is necessary to review the principles involved in these concepts. Such review is indicated in Appendix I.

Essential Features of the Redifon Equipment

(a) Use of Class 'D' amplification and pulse width modulation gives:

(i) High efficiency (50% overall, approx).

(ii) Following from (i), less cooling is required and consequently no fans are used and the transmitter is 'silent' in operation.

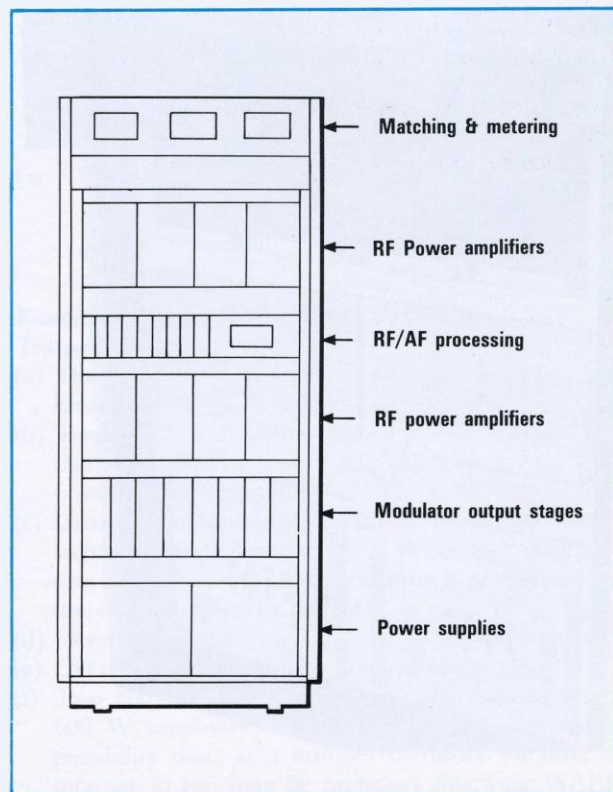(iii) Digital Circuits (whereby pulse techniques are used and devices are either in the 'on' or 'off' state).

Fig. 1. Basic schematic of Redifon MF transmitter.

Fig. 2. Basic 500 W or 1000 W transmitter.
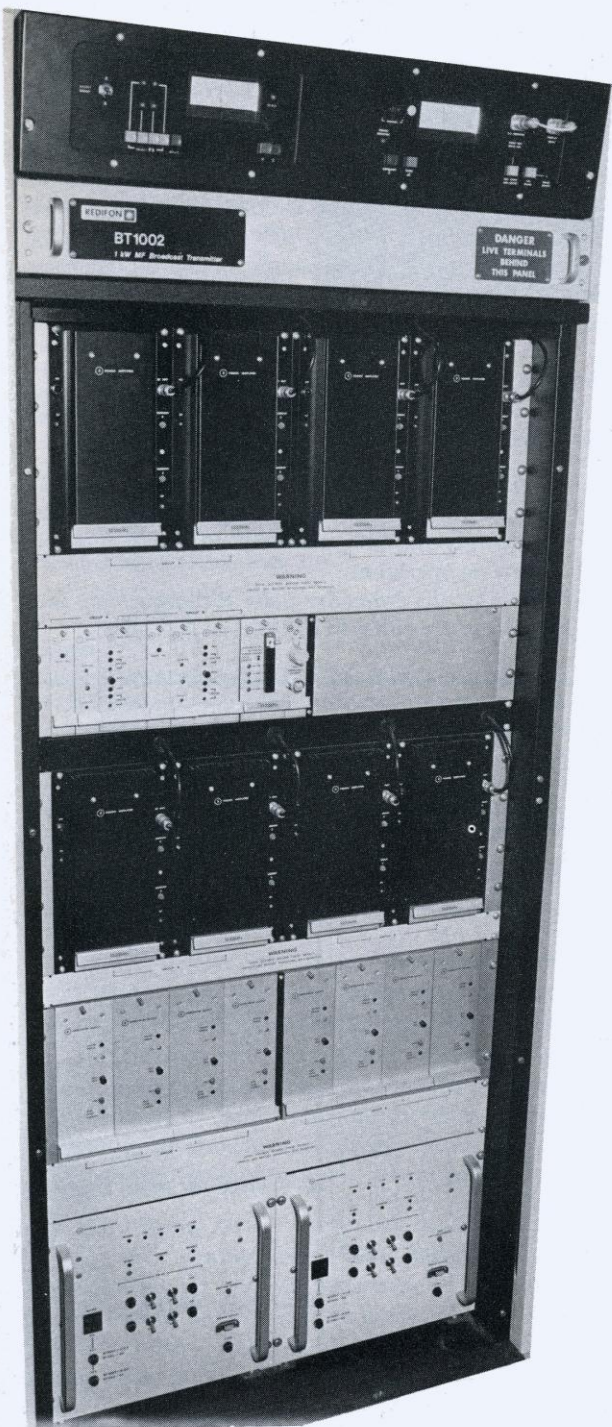
(iv) Modular Construction and Active Reserve Amplifier Configuration. (There are two RF drives in a passive reserve mode of operation).

(v) High reliability, as compared with designs based on thermionic valves, is anticipated. The relatively low heat dissipation will further improve reliability.

(b) Main Features of Technical Performance.

(i) Ease of power variation (pre-selected steps of 1, 2, 4 and 8 dB are available plus 'fine' continuous control).

(ii) Modulation capability 125%.

(iii) Harmonic Distortion
$\leq 2 \cdot 5\%$ up to 100% modulation depth.
Intermodulation Distortion 40 dB (two tone).

(iv) Hum and Noise (detected output)
60 dB below 100% modulation depth.

(v) Frequency Response
DC to 7·5 kHz ($-1\cdot5$ dB).

(vi) Spurious Outputs
60 dB below carrier level.

With special reference to the possibility of AM Stereo, the transmitter is capable of being phase modulated (incidental phase modulation $\leq 0\cdot1$ radian rms for 95% modulation).

NEW VHF/FM TRANSMITTER — BASIC CONCEPTS AND DESIGN

Introduction

The new VHF/FM transmitters (see Fig. 4), to be supplied by Harris Broadcast Products, are all solid-state broadband (87·5 to 108 MHz) in design. Each transmitter provides an output power of 300 W which is obtained from eight solid-state amplifier modules operating in parallel (see Fig. 5). The amplifier modules are driven from a 15 W broadband exciter (see Figs. 6 and 7) which contains a stereo encoder (switching type, which the manufacturers term a Digital Synthesised Modulator or DSM), a Synthesiser which can be set in increments of 50 kHz, RF amplifier and SCA generators. Two complete transmitters will be used in a full power passive reserve configuration.

There are a number of interesting concepts used in the design (Switching Type encoder, Overshoot Compensation, referred to by Harris as a Dynamic Transient Response filter, and the Synthesiser Design). A review of the basic principles involved in some of these aspects is included in Appendix I.
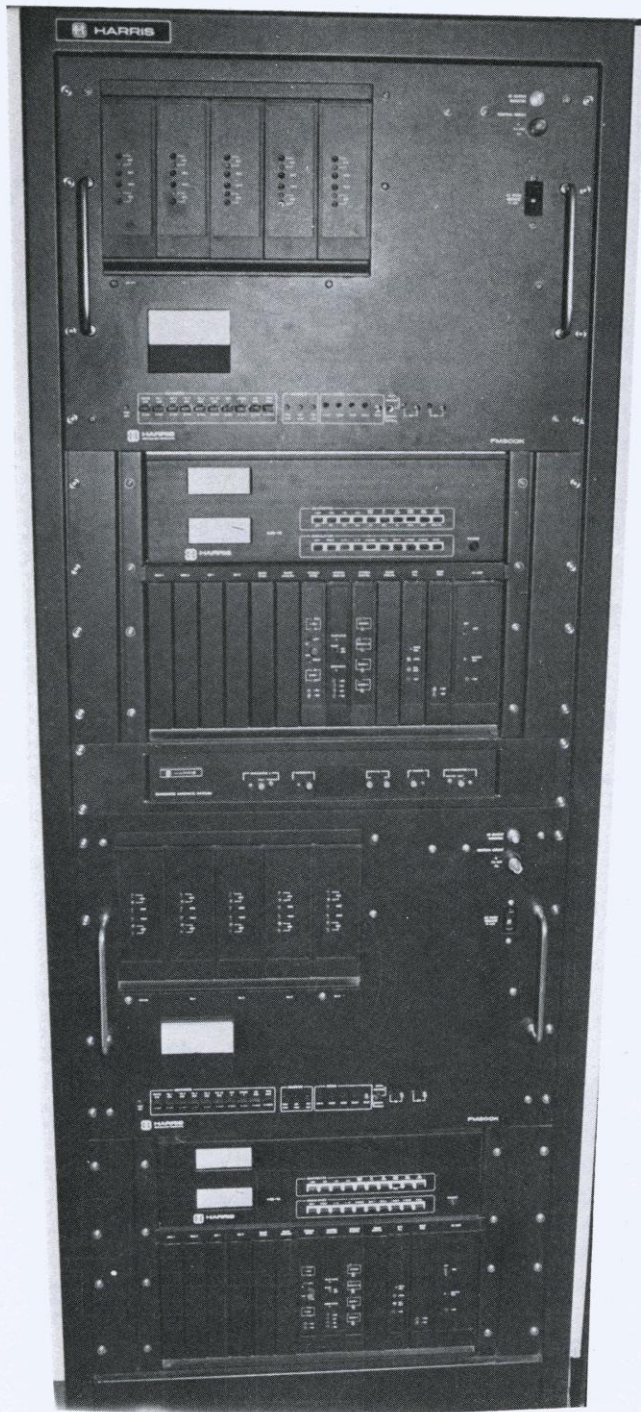

Fig. 3. BT 1002 transmitter.

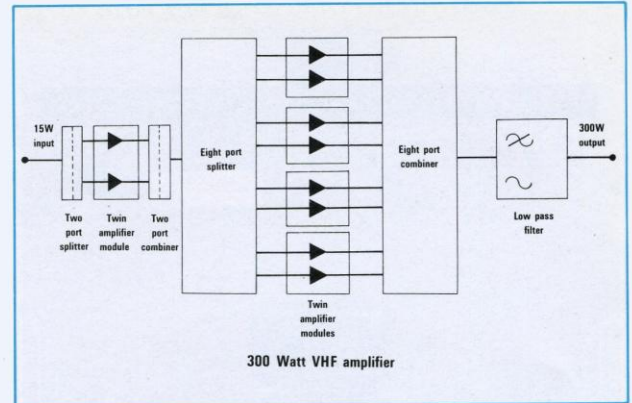Fig. 4. FM-300KD main/alternate 300 W FM transmitter.
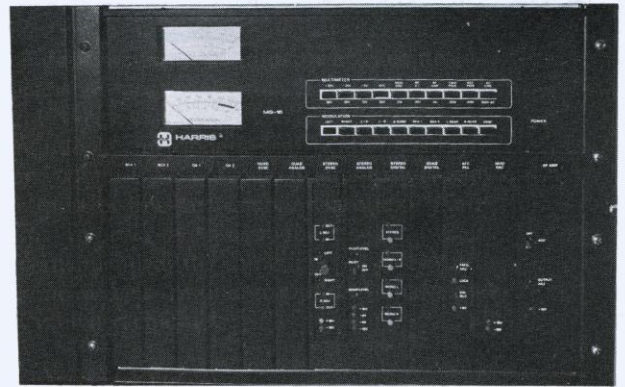

Fig. 5. 300 W VHF amplifier.

Fig. 6. MS-15 front view.

Essential Features of the 300W VHF/FM Transmitter

(a) The transmitter is generally modular in design and broadband 87·5 to 108 MHz.

(b) Two 300 W transmitter systems will be provided in a main and passive reserve configuration giving a full power standby capability.

(c) Great importance is placed on the Mono and Stereo technical performance of the ILR network. The IBA specification and expected transmitter performance is therefore as shown in Table 1.

(d) Stereo encoder is an integral part of the transmitter.

(e) Overshoot Compensation is provided.

(f) Two 300 W systems can be paralleled to provide a 600 W capability — this is useful because of the possibility that, at a number of future stations, an increase in erp may be necessary following WARC 1979.

Fig. 7. MS-15 rear view.

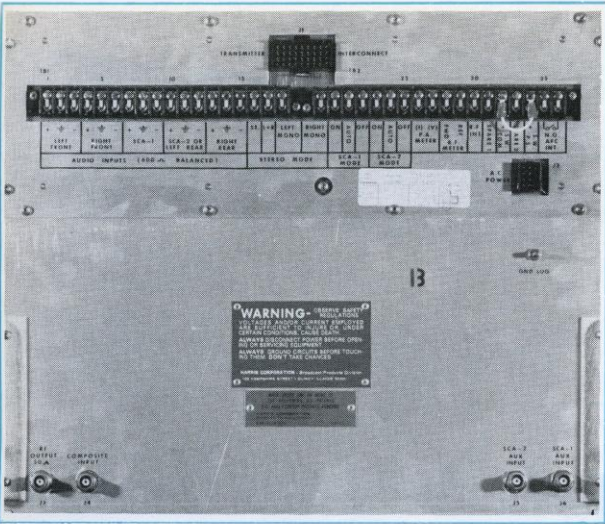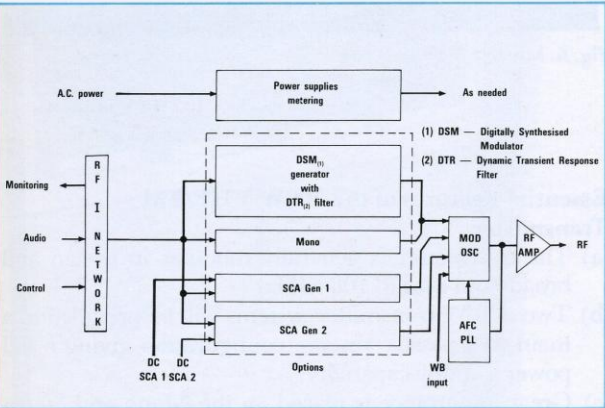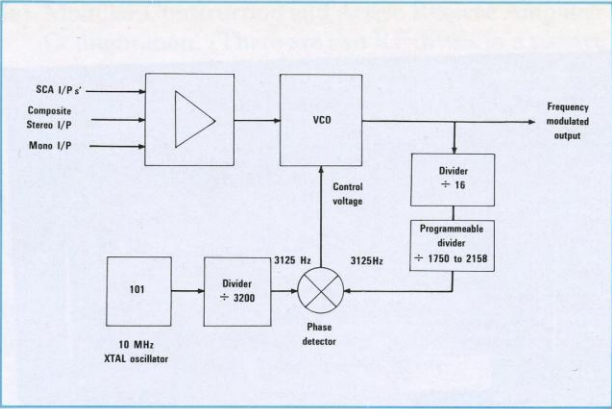**Fig. 9.** Modulated oscillator and AFC/PLL modules basic schematic.
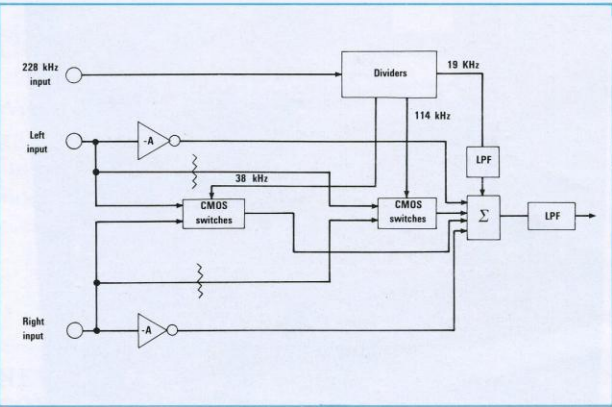

Fig. 8. Block schematic of MS-15.

Fig. 10. Block diagram of stereophonic digitally synthesised modulation (DSM).
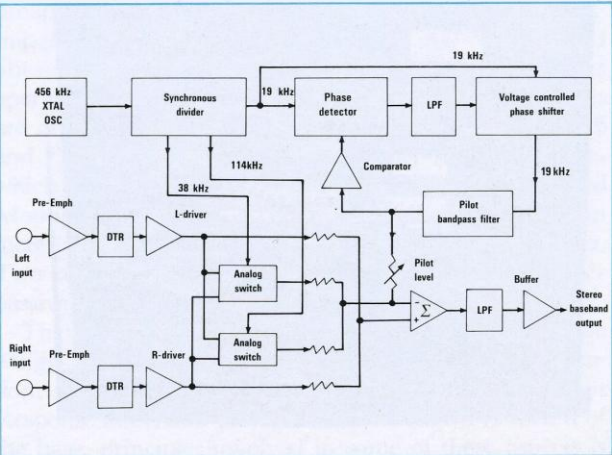

Fig. 11. DSM stereo generator block diagram.

TABLE 1: SUMMARY — VHF SPECIFICATION AND TYPICAL PERFORMANCE

Parameter	IBA Specification		Typical Harris 300 W Transmitter (Stereo Mode)
Audio Frequency Response	± 0·5 dB over range 40 Hz to 15 kHz		0·5 dB
Harmonic Distortion	30–125 Hz	0·6%	0·12%
	125 Hz to 7·5 kHz	0·4%	0·27%
	(at 75 kHz deviation)		
Audio Signal/Noise ratio			
Random FM Noise*	Unweighted	59 dB	64 dB
	Weighted	65 dB	69 dB
Random AM Noise		45 dB	70 dB
Unwanted AM due to FM	40 dB (below carrier when deviated 75 kHz)		60 dB

*Signal/Noise ratio has been expressed as a ratio of rms signal to quasi peak noise (measured at 75 kHz deviation, using a PPM).

Additional Parameters for Stereo:

Linear Crosstalk between left and right channels		
30 Hz– 4 kHz	38 dB	50 dB
4 kHz–15 kHz	Oblique segment rising at 6 dB/ octave 28·5 dB	48 dB

VHF AERIAL SYSTEMS

In the planning of the local radio network, emphasis has been placed on the needs of those listeners using portable transistor receivers and car radios. As a result, the transmission of a vertically polarised component of field, as well as a horizontal component to satisfy listeners with outside horizontally polarised aerials, has been specified. Special interest has been shown in the use of Circular Polarisation both for Phase I and Phase II stations. This paper discusses in some detail the concepts involved.

The maximum advantage to be gained from radiating a mixed polarised signal varies from between 6 dB for car reception and about 12 dB for reception at open area outdoor 'picnic' sites. However, it should be noted that in urban areas, and particularly inside houses, scattering of the signal can result in a randomly polarised signal such that little advantage is obtained.

In view of the foregoing, the specification for VHF aerials calls for the provision of circularly polarised aerial systems.

The required effective radiated powers range from 5 kW down to about 50 W with possibly even lower powers where only small areas are to be served. It should be noted that the erp of stations radiating signals of mixed polarisation is taken as the sum of those obtained from each plane of polarisation.

Summary of Typical VHF Aerial Specifications

|	
Horizontal Radiation Pattern (HRP)	Directional and Omni-Directional
Impedance:	50 ohms with voltage reflection co-efficient not exceeding 15% over the frequency band 94·5 to 97·5 MHz and 100 to 104 MHz and 10% at the operating channel.
Polarisation:	Left-hand Circular.
Polarisation Ratio:	±6 dB with the phase angle between the vertical and horizontal components to be within the range 90° and 45°.
System Gain:	Sufficient to achieve required erp.
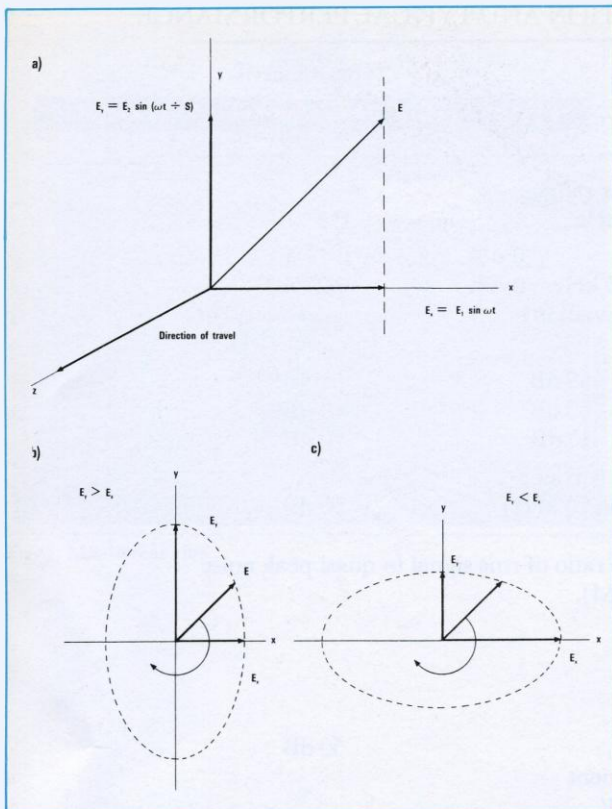
Fig. 12. An elliptically polarised wave may be represented by two mutually perpendicular linear waves as shown here.

Fig. 13. Slant and circularly polarised waves are special cases of the more general elliptical case.

TWINNED STATIONS

In the second phase of Independent Local Radio transmitting stations an additional concept has been introduced which has become known as the 'Twinned Station Arrangement'. This occurs where there are two areas which geographically are widely separated and, in consequence, provision of separate pairs of VHF/MF transmitters, to cover the areas individually, becomes necessary; for example, see Fig. 15 which indicates the probable arrangement for Dundee and Perth.

There are several options in respect of how the stations are operated from a programme viewpoint and these can be simply summarised as follows:

(a) A single studio with a single programme output to serve both areas.

(b) A single studio with a separate programming capability to the two areas.

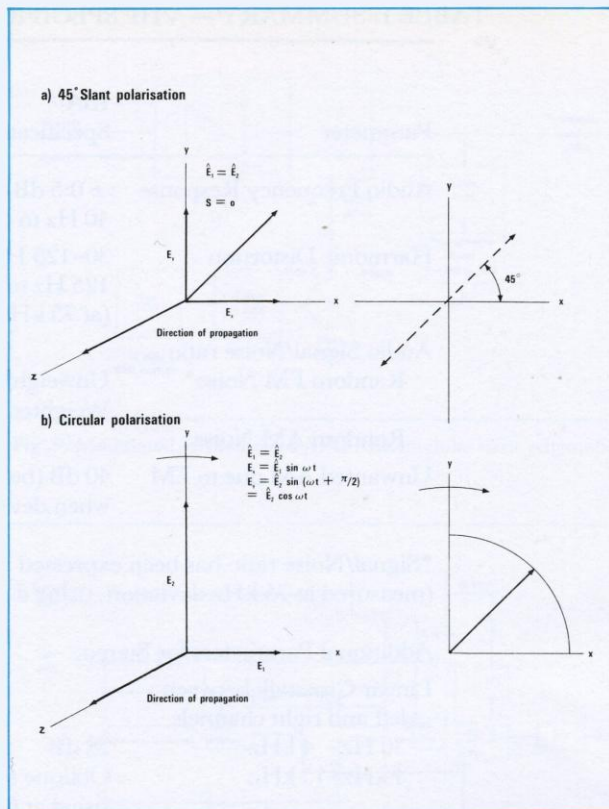(c) Two studios with the capability of common programming from the primary studio with the capability at the second studio being limited to feeding programmes to the second area only.

(d) Two studios, each with the capability of common programming and, of course, with each studio being able to feed exclusive programmes to its own transmitters. (See Fig. 16 which indicates the proposals for the Exeter/Torbay twin stations.)

At all these stations, certain difficulties which are related to the following will need to be overcome:

(i) The programme feed arrangements and, in particular, the problem of transmitting stereo programmes over relatively long distances; greater than 25 miles.

(ii) The monitoring of the transmitter in terms of quality is normally applied at the studios by means of an off-air receiver. This is not possible when the studio is an excessive distance from the transmitter as would be the case, for example, at Dundee where a single studio located in Dundee would be unable satisfactorily to receive the transmissions from Perth.
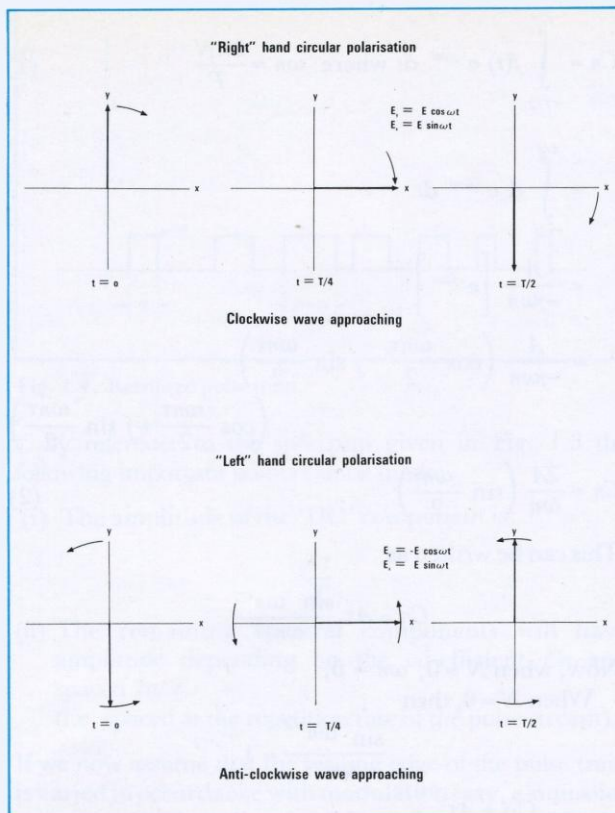
Fig. 14. Rotation of circularly polarised wave. The tip of the resulting vector of a circularly polarised wave traces a circular path. When viewed by an observer as an approaching wave, the rotation may be either clockwise or anti-clockwise (right-hand or left-hand), the direction of rotation being dependent on the relative phasing of the two mutually perpendicular waves representing the circularly polarised field.

Fig. 16. Exeter/Torbay twinned stations (tentative arrangement).

Another monitoring problem is the status of the transmitting station which is monitored by the Davis Over-Air Signalling System. Again, the problem is that of inability of receiving the over-air signal.

Various proposals as to how these problems might be resolved are currently being considered.

Acknowledgements

APPENDIX I: CLASS 'D' AMPLIFICATION AND PULSE WIDTH MODULATION

Class D Amplification

PULSE AMPLIFICATION

Class D amplification involves the processing of the information signal into pulse form, the amplification of these pulses, and the reconversion of the pulses back to the original information signal. By these means the amplifier, which is now amplifying only pulses, can be made highly efficient. Transistors can be used as switches, i.e., either on or off, and the theoretical efficiency of the amplifier approaches 100%.

The power dissipated in the transistor when it is turned ON is $V_{CESAT} \times I_{LOAD}$ and when it is turned OFF is $V_{CC} \times I_{CES}$. Both of these quantities are small compared with the power dissipated in the load R_1 which is:

$$\frac{V_{CC} - V_{CESAT}}{2R_1}$$

Fig. 15. Dundee/Perth twinned stations.

Fig. 1.1. Transistor used as switch.

Notice that the amplitude of the pulses is governed only by the supply rail voltage and that it is not possible to use this parameter of the pulse to convey information. The signal can only be transmitted by varying the timing of the pulses according to the information signal.
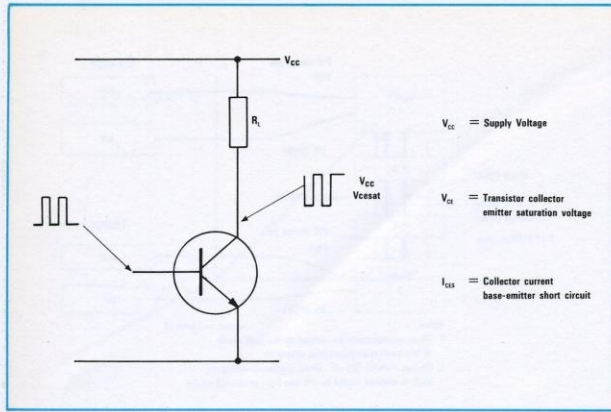
Pulse Width Modulation

In pulse width modulation the time duration of the pulses is modified by the modulating waveform. For the sake of analysis, if it is assumed that the leading edges of the pulse are independent of the modulation, and therefore occur at constant time intervals, and that the position of the trailing edge is modified in accordance with the modulation waveform, then an expression for the pulse width modulated signal can be derived. The mathematical analysis of a pulse width modulated system is complex; however, if one firstly considers a chain of rectangular pulses of fixed unmodulated width τ and of period T then an initial understanding can be obtained.

Consider the chain of pulses shown in Fig. 1.2 of width τ and of period T. The complex Fourier coefficient of this pulse train is given by:

$$Cn = \int_{-T/2}^{T/2} f(t)\, e^{-j\omega n t}\, dt \quad \text{where} \quad \omega n = \frac{2\pi N}{T} \qquad (1)$$

$$= \int_{-\tau/2}^{\tau/2} A\, e^{-j\omega n t}\, dt$$

$$= \frac{A}{-j\omega n} \left[e^{-j\omega n t} \right]_{-\tau/2}^{\tau/2}$$

$$= \frac{A}{-j\omega n} \left(\cos \frac{\omega n \tau}{2} - j \sin \frac{\omega n \tau}{2} \right) - \left(\cos \frac{\omega n \tau}{2} + j \sin \frac{\omega n \tau}{2} \right)$$

$$Cn = \frac{2A}{\omega n} \left(\sin \frac{\omega n \tau}{2} \right) \qquad (2)$$

This can be written as:

$$Cn = A\tau\, \frac{\sin \omega n^{\tau/2}}{\omega n^{\tau/2}}$$

Now, when $N = 0$, $\omega n = 0$,

When $N = 0$, then

$$\frac{\sin \omega n^{\tau/2}}{\omega n^{\tau/2}} = 1$$

$$\therefore\ Co = A\tau$$

Recall that $f(t) = \frac{1}{T} \sum_{n=-\infty}^{n=\infty} Cn\, e^{j\omega n t}$

$\therefore$ When $\omega n = 0$, the 'DC' component =

$$\frac{A\tau}{T}$$

The spectrum is shown in Fig. 1.3.
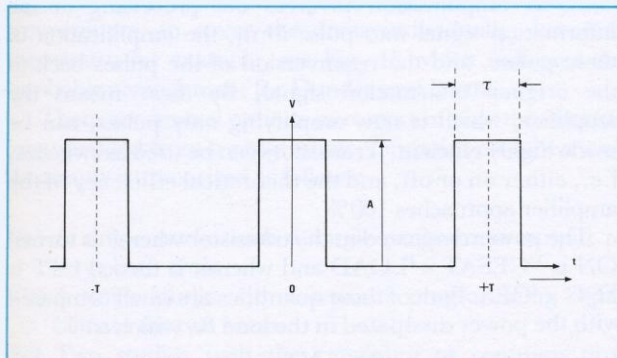
Fig. 1.2. Pulse width modulation (simplified analysis).
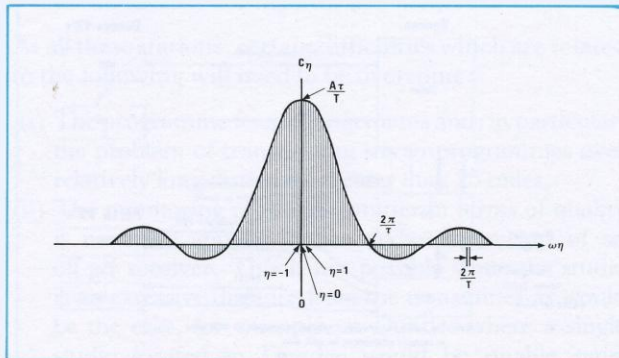
Fig. 1.3. Pulse width spectrum.
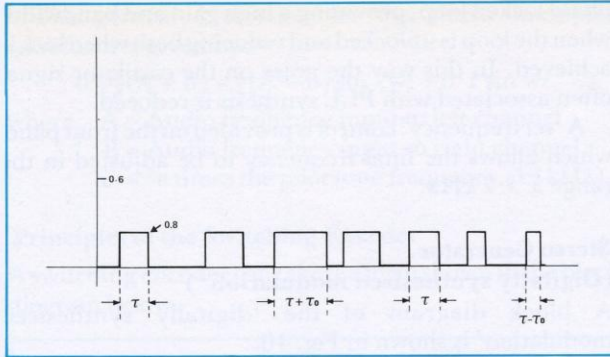
Fig. 1.4. Retrieved pulse train.

Fig. 1.5. τ varying in accordance with input voltage.

By reference to the spectrum given in Fig. 1.3 the following important points can be made:

(i) The amplitude of the 'DC' component is:

$$\frac{A\tau}{T}$$

(ii) The remaining spectral components will have amplitude depending on the co-efficient Cn and spaced $2\pi/T$
(i.e. spaced at the repetition rate of the pulse stream).

If we now assume that the trailing edge of the pulse train is varied in accordance with modulation (say, a sinusoid) such that the mean length of the pulse is τ and that the maximum and minimum length of the pulse is determined by a sinusoid $\tau_0 \sin pt$, such that the maximum is $(\tau + \tau_0)$ and the minimum $(\tau - \tau_0)$ (see Fig. 1.4), then the 'DC' component of the resulting spectrum is given by:

$$\frac{A\tau}{T} + \frac{A\tau_0}{T} \sin pt \qquad (1)$$

The complete expression for the spectrum can be derived (Reference 1) and is as follows:

$$f(t) = \frac{A}{T}\left[(\tau - \tau_0 \sin pt)\right] + 2\sum_{K=1}^{\infty}\frac{1}{K}\sin(Kwt)$$

$$-2\sum_{K=1}^{\infty}\sum_{n=-\infty}^{\infty}\frac{1}{k}Jn(Kw \times \tau_0)\sin\left[(Kw + np)t - Kw\tau\right] \quad (2)$$

The first part of expression (2) is precisely as given in equation (1) and contains a term at the modulation frequency $\sin pt$, whose magnitude is dependent on $A\tau_0/T$, i.e. proportional to the amplitude of the pulse and τ_0, the depth of modulation.

The undistorted modulation waveform can, therefore, be retrieved by passing the pulse train through a low-pass filter which removes the unwanted components and their

sidebands which are at multiples of the pulse repetition frequency (w).

Reference
1. R. D. Stuart. An Introduction to Fourier Analysis. Methuen's Monographs on Physical Subjects.

CLASS 'D' AUDIO AMPLIFIER

If we precede the pulse amplifier of Fig. 1.1 by a pulse width modulator (a device which varies the width of the pulse τ, linearly depending on an incoming signal) and follow it by a low-pass filter, then the 'DC' output of the filter will vary in a direct linear relationship with the input signal, i.e. we have a DC amplifier. By allowing the 'DC' to vary, but at a much slower rate than the clock pulses, we can obtain a Class D audio amplifier. A suitable clock rate would be 10 times the highest audio frequency.

Fig. 1.6 illustrates how the above techniques are used in the Redifon Transmitter. A pulse width modulator

Fig. 1.6. Schematic of Redifon modulator and output stage.

using the ramp comparison technique, is used to provide high level audio to an RF power amplifier. The RF amplifier comprises four transistors in a bridge configuration switching at RF across the audio modulated supply. Since the RF transistors are either 'on' or 'off' the dissipation is low.

APPENDIX II

Circuit Description

Electrically the MS15 consists of four sections; the modulator-driver, audio processing, power supply and the monitoring circuits. The modulator-dri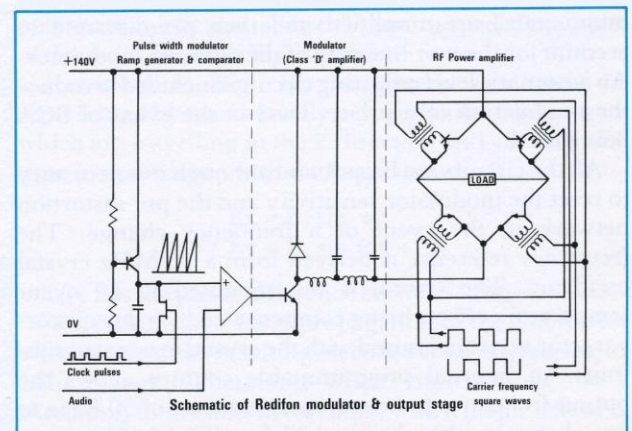ver is further subdivided into three modules, the modulator-oscillator, the AFC and the RF amplifier. Similarly, the audio processing is subdivided into an overshoot compensator module and a stereo encoder consisting of an analogue module and a digital module. A block diagram of the complete equipment is shown in Fig. 8.

Modulator-oscillator and AFC/PLL Modules

A block diagram of the modulator oscillator and AFC/PLL modules is shown in Fig. 9. The VCO is a varactor controlled Hartley oscillator housed in a screened box, some components inside this unit being encapsulated in a transparent resin. The active element in the oscillator is a MOSFET, as is the RF buffer amplifier contained in the sub-unit. A broadband RF amplifier raises the power level to 250 mW and also provides a second output of 2 mW for the phase locked loop.

The baseband input signals (stereo composite, SCA, mono, etc.) are amplified and then pre-distorted to account for the non-linearity of the varactor modulator. An automatic level adjusting circuit is included to reduce the modulation to a pre-set level in the event of SCA being used.

All the circuits are broadband although it is necessary to reset the modulator sensitivity and the pre-distortion network in the event of a frequency change. The frequency reference is derived from a 10 MHz crystal oscillator. The crystal is not mounted in an oven, temperature effects being compensated by a thermistor-varactor network housed with the crystal in a sealed sub-unit. An internal programmable counter allows the output frequency to be set, in increments of 50 KHz to any channel within the band 87·5 to 107·9 MHz.

Two filters are used to determine the bandwidth of the phase locked loop, providing a high gain and bandwidth when the loop is unlocked and reducing both when lock is achieved. In this way the noise on the oscillator signal often associated with PLL synthesis is reduced.

A 'set frequency' control is provided on the front panel which allows the final frequency to be adjusted in the range ± 1·7 kHz.

Stereo Generator
(Digitally synthesised modulation*)

A block diagram of the 'digitally synthesised modulation' is shown in Fig. 10.

The left and right inputs are fed, via an RFI filter network, to balanced transistor input amplifiers which are used in preference to transformers.

The pre-emphasis circuits in the input amplifier are selected by on-board plug links and cannot be switched out either from the front panel or remotely. The mode of transmission (stereo, mono source left, mono source right, mono source left and right) can be selected remotely or by using the front panel controls (on the DSM digital module), the switching elements being CMOS switches.

The stereo encoder itself is of the switching type, the output being switched at 38 kHz between the left and right channels and the pilot tone added in a summing amplifier. To ease the filtering requirements an appropriate amplitude sample of the inputs, switched at 114 kHz (third harmonic), is also added, the phase being adjusted to be +180° with respect to the existing third harmonic. Thus, the first harmonic requiring filtering is the 5th (190 kHz) as against the 3rd 114 kHz. The switching elements are again CMOS switches.

Some of the advantages of using a switching type stereo encoder are as follows:

(1) In a conventional balanced modulator arrangement, it is difficult to maintain a true 'balance' condition due to temperature and ageing effects.

(2) A balanced modulator suffers from having a number of interactive adjustments.

(3) Any non-linearity in a balanced modulator causes distortion to the (L−R) sidebands and also generates harmonics which interfere with the SCA carrier (i.e. spurious 2nd order sideband components around 76 kHz).

The basic principles involved in a switching type encoder are outlined below.

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*The term 'digitally synthesised modulation' is used by Harris to describe their stereo generator which is based on the switching type encoder.

The general expression for the stereo multiplex waveform is given by:—

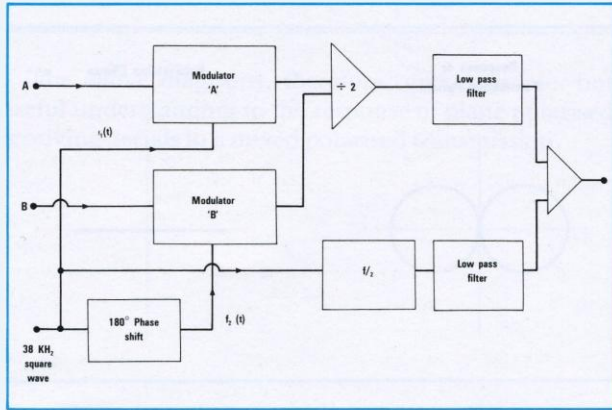$$0 \cdot 9 \tfrac{1}{2}(A + B) + \tfrac{1}{2}(A-B) \sin 2wt + 0 \cdot 1 \sin wt$$

where A = Audio frequency input to left channel
B = Audio frequency input to right channel
ω = $^2\pi$ times the pilot tone frequency (19 kHz)

Principles of the Switching Encoder

A switching encoder can take the form shown in the block diagram below.

The subcarrier signals are given by:

$$f_1(t) = 1 + \sin 2wt + \tfrac{1}{3}\sin 3(2wt) + \tfrac{1}{5}\sin 5(2wt) + \ldots$$

$$f_2(t) = 1 + \sin(2wt + \pi) + \tfrac{1}{3}\sin 3(2wt + \pi) + \tfrac{1}{5}\sin 5(2wt + \pi) + \ldots$$

where sin wt = Pilot tone (19 kHz)

Expanding $f_2(t)$

$$= 1 + (\sin 2wt \cdot \mathrm{Cos}\,\pi + \mathrm{Cos}\,2wt \cdot \sin \pi)$$
$$+ \tfrac{1}{3}(\sin 6wt \cdot \mathrm{Cos}\,3\pi + \mathrm{Cos}\,6wt \cdot \sin 3\pi)$$
$$+ \tfrac{1}{5}(\sin 10wt \cdot \mathrm{Cos}\,5\pi + \mathrm{Cos}\,10wt \cdot \sin 5\pi) + \ldots$$

Simplifying,

$$= 1 + (-\sin 2wt + 0) + \tfrac{1}{3}(-\sin 6wt + 0)$$
$$+ \tfrac{1}{5}(-\sin 10wt + 0) + \ldots$$
$$= 1 - [\sin 2wt + \tfrac{1}{3}\sin 6wt + \tfrac{1}{5}\sin 10wt + \ldots]$$

The output of Modulator (A) is given by:

$$A + A[\sin 2wt + \tfrac{1}{3}\sin 3(2wt) + \tfrac{1}{5}\sin 5(2wt) + \ldots]$$

and the output of Modulator (B) is given by:

$$B - B[\sin 2wt + \tfrac{1}{3}\sin 3(2wt) + \tfrac{1}{5}\sin 5(2wt) + \ldots]$$

The output of the ($\div 2$) summing amplifier is therefore:

$$\frac{(A + B)}{2} + \frac{(A-B)}{2}[\sin 2wt + \tfrac{1}{3}\sin 3(2wt) + \tfrac{1}{5}\sin 5(2wt) + \ldots]$$

filtering out the components above ($2wt$):

$$= \frac{(A + B)}{2} = \frac{(A-B)}{2}\sin 2wt$$

It should be noted, therefore, that if the 3×2 wt (114 kHz) is cancelled out, the restraints placed on the lowpass output filter design are considerably eased.

All that is now required is the addition of the pilot tone and the composite stereo signal has been obtained. The modulators themselves are, at any instant, either passing the signal unchanged, or completely stopping it. These modulators can, therefore, be realised in the form of switches.

Also contained in this module is a 17·5 kHz low pass filter prior to the switches. Although this filter can operate in isolation, it is normally part of an overshoot compensation system.

Digital Module of the Stereo Generator

The DSM digital module performs two functions; to generate the various switching frequencies required by the DSM analog module, and to provide interface and controls for the mode switching (see Fig. 11 which gives complete block diagram of DSM stereo generator).

APPENDIX III

BASIC THEORY OF CIRCULAR POLARISATION

Basic Theory

It is necessary to firstly consider elliptical polarisation, since it is convenient to regard the other types of polarisation, circular, slant or linear, as special examples of the more general elliptical case.

Referring to Fig. 12 the general case may be represented by two mutually perpendicular linear waves which are travelling in the Z direction and varying with time in accordance with the expressions:

$$Ex = E_1 \sin \omega t$$
$$Ey = E_2 \sin (\omega t + S)$$

where S is the phase difference between the two waves.

This results in a wave propagation in the Z direction with the tip of its resulting electric vector, E, tracing an elliptic path. Therefore, when viewed from the Z direction, this results in an ellipse with a major axis of E_1 or E_2, depending on the relative magnitudes. It is now possible to consider some examples of specific interest,

such as 45 degree slant polarisation and circular polarisation.

These are indicated in Fig. 13 and occur when $E_1 = E_2$, $S = O$ and $E_1 = E_2$, $S = \pi/2$ respectively.

Of course, some other special cases occur when $Ey = O$, which results in horizontal polarisation, or when $Ex = O$, which results in vertical polarisation. It should be noted from the above that a circular or slant polarised field can be produced by using two linearly polarised aerials, one radiating vertical and the other horizontal polarisation in an appropriate phase relationship.

One further factor that is important, and should be discussed, is the direction of rotation of the wave. By reference to Fig. 14 where a circularly polarised wave is illustrated with $Ey = E \cos \omega t$, and $Ex = E \sin \omega t$, it can be seen that the rotation of the wave is clockwise in the positive Z direction (with wave travelling out of page). This is classically referred to as right hand circular polarisation. If Ey is reversed, $(= -E \cos \omega t)$, then left hand circular polarisation is obtained.

Therefore, the following definitions can be made:

(a) Right hand circular polarisation occurs when the wave rotation is clockwise approaching, or counter-clockwise receding.

(b) Left hand circular polarisation occurs when the wave rotation is anti-clockwise approaching, or clockwise receding.

It should be noted that these definitions are based on classical physics usage and, as already stated, the Authority has adopted left hand circular polarisation for its VHF service.

General Analysis of an Elliptically Polarised Wave received by a Plane Polarised Receiving Aerial.

By reference to the figure below, the field induced in a dipole orientated at an angle θ by an elliptically polarised wave is proportional to:
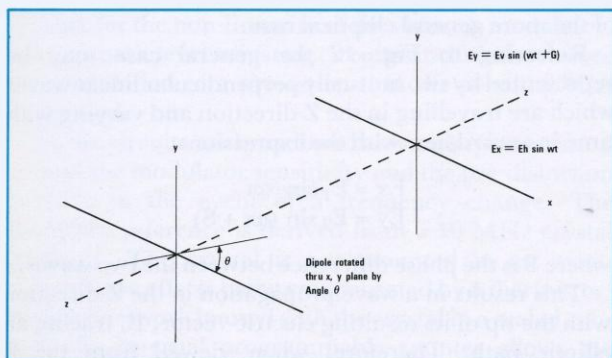
$$E(\theta) = K(E_H \sin wt (\cos \theta) + Ev \sin (wt + S) \sin \theta)$$ where K is a constant.

From this expression, therefore, the response (or polarisation pattern) can be obtained for the dipole from the various special cases of elliptical polarisation (circular, slant etc.). These can be demonstrated as follows:

Horizontal Plane Polarisation

$Ey = O$
$Ex = Eh \sin wt$
$E\theta = Eh \sin wt (\cos \theta)$
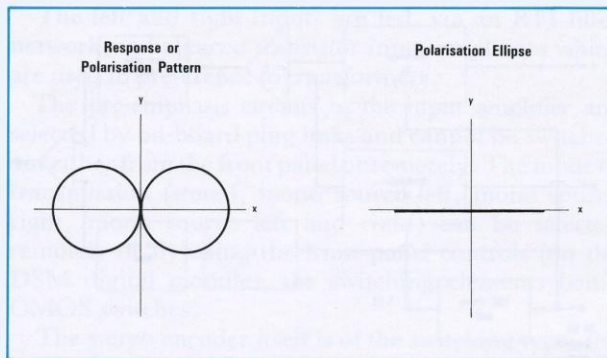
Slant

$Ey = Ev \sin wt \ (S = O°)$
$Ex = Eh \sin wt$
$Eh = Ev$
$E\theta = k(Eh \sin wt (\cos \theta) + Ev \sin wt (\sin \theta))$

Circular Polarisation (R.H.)

$Ex = Eh \sin wt$
$Ey = Ev \sin (wt + 90°), S = 90°$
$Eh = Ev$
$E\theta = k(Eh \sin wt (\cos \theta) + Ev \cos wt (\sin \theta))$

Response or
Polarisation Pattern

Polarisation Ellipse

1.0 KE
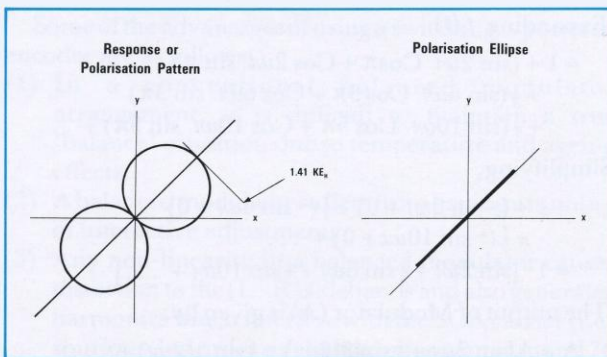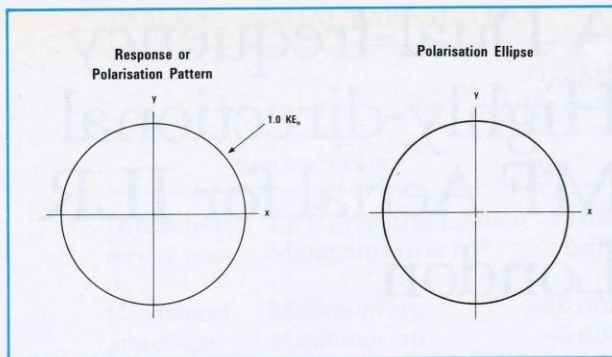
The above diagrams, therefore, give a simple, but useful understanding to the response of plane polarised receiving aerials to a mixed polarised transmission.

TED FORD joined the IBA in 1972 and is currently Principal Engineer in the Masts and Aerials Section of Station Design and Construction Department. His responsibilities include the engineering of MF and VHF transmitting aerials for Independent Local Radio and the provision of channel combining equipment for UHF Television's Fourth Channel. He obtained a BSc (Eng) degree in 1963 and worked for four years on microwave radar aerials and components at Associated Electrical Industries in Leicester, followed by four years on VHF and microwave aerials for spacecraft at the European Space Research Organisation, before joining the IBA.
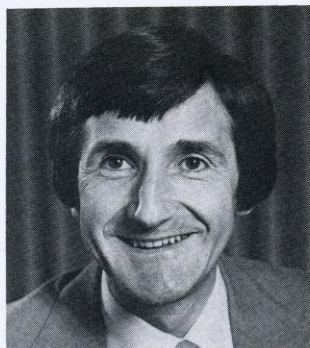
A Dual-frequency Highly-directional MF Aerial for ILR London

by E T Ford

Synopsis

The directional MF transmitting aerial at the ILR London station, at Saffron Green near Borehamwood, is technically the most complex MF aerial system built by the IBA. Not only does it radiate two programme services simultaneously towards London, but also it aims deep nulls towards ten cities that are served by other ILR and BBC stations sharing the same channels.

The degree of directivity required by the radiation patterns is as high as that achieved by the Authority's four-mast aerial arrays at Birmingham and Manchester, described in *IBA Technical Review 5*. The special requirement of dual-frequency operation at London introduced problems of channel-separation filtering.

INTRODUCTION

In 1971 the IBA planned to use a dual-frequency aerial array for its London Independent Local Radio Services on 1151 kHz and 1546 kHz (changed in November 1978 to 1152 kHz and 1548 kHz) and employed North American consulting engineers to design the aerial in a form suitable for supply and commissioning by UK contractors. With ten co-channel stations to protect in the UK — five on each frequency — it was required that groundwave nulls down to 24 dB at various angles to the north and west be provided. The technology for achieving stable nulls at chosen directions and depths from highly-directional arrays was well established in the USA and Canada for single-frequency operation, and requires no further comment here. However, owing to the nature of broadcasting in North America, the sharing of two or more services on a single aerial was unusual, and experience in this field was therefore limited.

The reasons for the IBA decision to co-site the two services were those of the cost-saving advantages of sharing an aerial system and transmitter building on a common site and the expected difficulty of obtaining a suitable alternative site in the London area. Only when a site had finally been acquired at a location 19 km north of the city centre could the aerial parameters be fully defined (see Table 1).

The services were to be broadcast 24 hours each day, and with local groundwave coverage only, the radiation patterns being unchanged day and night. Different radiation patterns were required at each frequency to meet the groundwave specifications listed in Table 1, which show that at 1151 kHz the most significant problem would be that of establishing a stable null at least 24 dB below the main lobe over the 15-degree angular arc from 301° to 316° towards Birmingham, only 145 km distant northwards. At 1546 kHz the main problem would be in establishing a level of at least 20 dB below the main lobe towards Bristol, 164 km distant, lying on the skirt of the main lobe only 100° off the direction of maximum radiation. Skywave and groundwave radiation had to be suppressed, to a lesser extent, towards the other co-channel stations on each frequency

TABLE 1— AERIAL PARAMETERS

		1152 kHz (formerly 1151 kHz)		1548 kHz (formerly 1546 kHz)	
	SPECIFICATION	erp	DIRECTION	erp	DIRECTION
Groundwave service area	ERP to central London	14 dBkW	160° ETN	+ 20 dBkW	160° ETN
	Minimum erp at 80°	0 dBkW	80° to 240°	+ 5 dBkW	80° to 240°
Co-channel protection	Maximum erp	−10 dBkW	301° to 316°	0 dBkW	261° to 265°
	Maximum erp	−5 dBkW	324° to 348°	+ 5 dBkW	315° to 348°

An erp of 0 dBkW represents a field strength of 300 mV/metres at one kilometer from the aerial, under loss-less propagation conditions.

(the skywave erp being limited generally to not more than −5 dBkW at 1151 kHz and 0 dBkW at 1546 kHz, in the appropriate directions). This made necessary the choice of an end-fire aerial array with four masts. All four of these would need be fed at each frequency with independent control of the phases and amplitudes of each radiator in order to steer the nulls into the required amplitudes and directions.

THE AERIAL DESIGN AND INSTALLATION
The site area allowed provision of an array of four guyed masts equally spaced by 61 m along a line oriented at 161° east of north. Each steel lattice mast is 71 m high, with a triangular cross-section of 430 mm face width, and has a ground plane consisting of 120 buried wires extending to a radius of 65 m.

In the horizontal plane the width of the radiation pattern main lobe is to a large extent governed by the compromise choice of intermast spacing. The half-power beamwidth at (currently) 1152 kHz is 90°, whereas at (currently) 1548 kHz it is 114° because the mast spacing is equivalent to slightly more than 0·3 wavelength. In consequence, the aerial gain at 1548 kHz is about 1 dB lower. The combination of lower gain with the significantly higher groundwave propagation attenuation at the higher frequency necessitates a large difference between the transmitter powers to achieve the same service area on both channels. At present, the system is operating with transmitter powers of 27·5 kW at 1548 kHz but at only 5·5 kW at 1152 kHz. The erp and aerial gains measured under these conditions show that the overall aerial efficiency is about 84% at both frequencies (see Table 2).

Dual-frequency operation requires that the channels be isolated from each other for two different and unrelated purposes. One is the prevention of cross-modulation and intermodulation between the transmitters which, if these were the only problems present, could be solved simply by the fitting of a suitably rated rejection filter at each transmitter port. The other is the isolation of the matching and power-dividing networks, necessary to avoid upsetting the amplitudes and phases of the mast radiators at one frequency while the networks at the other frequency are being tuned.

For independent non-interactive control of two aerial patterns with critical null requirements it was considered essential to fit rejection filters between each mast and its associated tuning network.

Therefore, eight coaxial transmission lines are employed to separately carry the power from the two transmitters to the four masts, and combining is done directly at each mast base through pass/reject filters, each comprising a single series/parallel resonant circuit in series with the line. The filters also provide sufficient isolation between the transmitters, thus obviating the need of more filtering at the transmitter ports.

Since, by international standards, the transmitter powers are fairly modest, the system employs principles of network design and transmission-line layout similar to those of the IBA low-power single-frequency directional aerials described in *IBA Technical Review 5* (Ref. 1). The only difference is in the higher ratings of components, which are designed for 30 kW at 1548 kHz and 10 kW at 1152 kHz using numerous ceramic vacuum capacitors.

ISOLATION BETWEEN TRANSMITTERS
The rejection of unwanted energy from a transmitter depends not only on the rejection afforded by the filters but also on additional rejection by reason of the matching, phasing, and power-splitting circuits associated with the other frequency being off-tune. In a multi-mast array this additional rejection is largely unpredictable; for example, the overall rejection of the carrier and sidebands obtained at 1548 kHz is much higher than is obtainable at 1152 kHz. However, since

TABLE 2 — ERP AND AERIAL GAINS

MEASURED PARAMETERS	1152 kHz (formerly 1151 kHz)	1548 kHz (formerly 1546 kHz)
Aerial Maximum Intrinsic Gain	+ 7·1 dB	+ 6·2 dB
Losses in Main Feeders	−0·2 dB	−0·2 dB
Losses in Aerial Networks and Earthmat	−0·6 dB	−0·5 dB
Transmitter Power	+ 7·4 dBkW	+ 14·4 dBkW
Net erp on Aerial Main Lobe	+ 13·7 dBkW	+ 19·9 dBkW

the low-frequency transmitter is 7 dB lower in output power, the net result is that the isolation through the aerial circuits, together with the mixing-loss of the Marconi B.6029 transmitters, has proved adequate to suppress cross-modulation and inter-modulation.

RE-RADIATION PROBLEMS
Directional aerials impose restraints upon the size and orientation of suitable sites; and, in any urban environment, there might be very little choice as to their locations. Consequently, it is probable that any chosen site will be close to other re-radiating structures such as tall buildings or electricity supply pylons.

At London, the site was adjacent to two lines of electricity pylons to the north and east, the pylons being of various heights between 27 m and 32 m and about 500 m distant at their closest approach. The situation was complicated by the presence of a 24 m steel tower at a distance of 700 m in front of the aerial in the main lobe.

Earlier IBA directional aerial installations had been subject to signal reflections from pylons, but the London case was the most severe. In addition, pattern specifications at two frequencies had to be met. In North America, pylon detuning at single frequencies is often undertaken, in collaboration with the power companies, and is a very time-consuming process. The London installation presented a dual-frequency problem, and investigation was made with a view to avoiding recourse to detuning.

The first step was a computer study of the probable effects of the pylon re-radiation. The mutual coupling between the aerial and the nearest 27 pylons and the tower was calculated (by using, for simplicity, theoretical directional radiation patterns of the aerial, and the assumption that all the pylons were of equal height). The radiation patterns from this giant array of 32 radiating sources (which included the four aerial masts) were then computed, the radiation being assumed to emanate from the vertical pylon structures and not from the horizontal overhead wires. Correlation with some measured results

revealed that an approximate simulation could be obtained by assuming the magnitude of base self-impedance of the pylons as being between 200 and 300 ohms, dependent on frequency. The phase angle of this impedance was uncertain, so several phase conditions were explored to assess the effect upon groundwave and skywave radiation.

The predictions showed that, regardless of the phase of the re-radiated energy, the resulting aerial patterns in the null regions would be broken by as many as a dozen narrow lobes at both frequencies. This afforded insight into the probable limitations in the angular areas over which nulls at specified depths in the groundwave and skywave patterns could be established. Calculated patterns for the edges of the audio bandwidth also were made; and, predictions showed that, despite the extended layout of the pylons, the pattern bandwidth problems would be negligible and the required null depths could be obtained without detuning the pylons.

AERIAL COMMISSIONING
The aerial commissioning was undertaken as a joint effort between the main contractor, the consulting engineers and IBA staff.

The object was of finding the minimum back-lobe levels achievable without detuning any pylons. At the higher frequency, which at that time was 1546 kHz (changed to 1548 kHz in November 1978), null levels relative to the main lobe, of −23 dB towards Bristol, and −17 dB towards other co-channel stations within the arc 315° to 350°, were achieved and found to be acceptable. Further improvement presented difficulty and these results were much like those predicted in the computer analysis.

At the lower frequency, which was then 1151 kHz, levels of −23 dB were initially obtained within the arc 300° to 350°. However, it was critical that, within the arc 301° to 316° towards Birmingham, a level of −24 dB be established. It was found that, by further adjustment of the currents on the aerial masts, a null of −35 dB towards
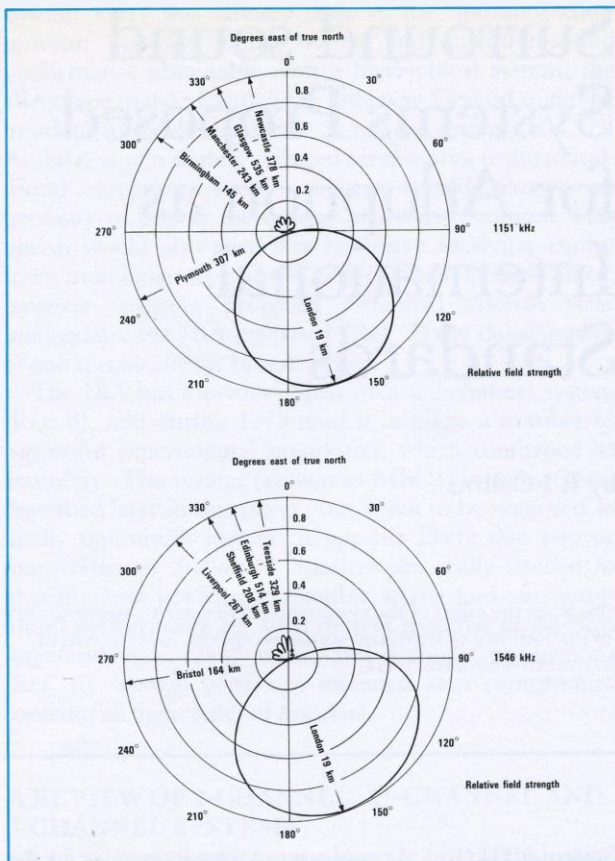
Figs. 1a and 1b. Measured Radiation patterns, 1152 kHz and 1548 kHz.

central Birmingham, and of −24 dB at the extreme edges of the arc, could be generated. No pylon detuning was necessary, and this arrangement is still maintained.

To measure the groundwave patterns, field strength was plotted against range on 16 bearings, at ranges typically from 1 km to 20 km. This necessitated measuring at about 240 locations. To obtain valid information about the depths of the nulls within the arc 260° to 350°, measurements were made at ranges of at least 10 km, ie, at selected points ten times further from the aerial than those pylons which contributed in larger part to the re-radiation. The Birmingham null was investigated as far as Birmingham itself, 145 km distant. The measured patterns are shown in Figs. 1(a) and 1(b).

MAINTENANCE OF THE LONDON AERIAL

At London, the IBA services are broadcast 24 hours per day; so, the policy for routine maintenance is of having a reserve mode of transmission that is, so far as possible,

independent of the mechanical and electrical structure of the aerial. The reserve aerial consists of a wire radiator sloping at 45° from the top of the south mast on its south side. Erection of the aerial occupies only a few minutes, during which time transmissions are interrupted.

The bases of all four aerials are then short-circuited to earth, and the wire radiator is fed at its base via independent coaxial feeders and a matching/combining network. The masts act as passive reflectors, resulting in the production of broad cardioid patterns towards London. The transmitter powers are reduced to 10 kW and 4 kW respectively, with the result that the co-channel protection to the north and west, and the erp to the south, are degraded by no more than 10 dB. Most of the 3 mV/m service area contour is then degraded to a 1 mV/m contour, although some areas near the sides of the main aerial lobes are unaffected because the cardioid lobes are broader.

The transmitters being thus disconnected from the main aerial networks, and the masts being earthed, mast and aerial maintenance work can proceed with safety. Also, the RF networks can be inspected, disconnected or repaired, without hazard.

SUMMARY

Since March 1975, the London station has provided 24-hour daily broadcasting services. Throughout that period the field strengths in the main lobe have been monitored, and the critical null regions of the aerial patterns have consistently been sufficiently stable to remain within the specified limits. There has been no report of co-channel interference with any other UK broadcaster, nor of any significant cross-modulation or intermodulation between the two IBA services.

The aerial efficiency is satisfactory and the required erp and service-area coverage have been achieved.

The reserve mode of operation for maintenance purposes under 24-hour broadcasting conditions has been successfully demonstrated.

ACKNOWLEDGEMENTS

Reference
1. Ford, E T — 'Directional MF Aerial Arrays for the Independent Local Radio Service', *IBA Technical Review 5, p. 45.*

R. I. COLLINS joined the IBA in 1968 as a graduate engineer in the Radio Frequency Section of the Experimental and Development Department. He worked on development of receivers and demodulators. Recent work has included research and development in Surround-sound and radio projects for the Independent Local Radio service. Ian Collins is married and lives in Winchester.

Surround-sound Systems Proposed for Adoption as International Standards

by R I Collins

Synopsis

This article provides a review of 2-channel, $2\frac{1}{2}$-channel and 3-channel surround-sound systems. The specification of the IBA 3-channel system MSC-1 for broadcasting is given together with the basic characteristics of the system. The author discusses the effect of the system on service areas, susceptibility to interference of surround-sound reception, and the cost of surround-sound receivers.

INTRODUCTION

In 1976 the IBA commenced a study of the existing proposals for surround-sound broadcasting. These included several 2-channel matrix systems and the NRDC/Ambisonics System 45J (Ref. 1), which was hierarchical (ie, the same broadcast can be received in the 3-channel, $2\frac{1}{2}$-channel or 2-channel mode). The IBA found that the surround-sound performance of 2-channel systems was inadequate for some types of material, whereas $2\frac{1}{2}$-channel System 45J gave good results. A theoretical study (Ref. 2) showed this matrix system to have highly favourable properties relating to coverage areas in both stereo and surround-sound reception.

The IBA therefore concluded that this $2\frac{1}{2}$-channel system should be developed for broadcasting, and that experimental transmissions should be made. It also concluded that signals should be originated by using Ambisonic studio technology (Ref. 3), in which horizontal surround-sound signals are processed in a 3-channel format ('B Format'). No significant advantage was found in using more than three channels for horizontal surround-sound.

Meanwhile, System 45J was slightly modified by NRDC/Ambisonics to yield a new hierarchical matrix,

System UHJ (Ref. 4), and a set of tolerance zones for the basic 2-channel encoding (known as 'HJ') was agreed between the NRDC and the BBC (Refs. 4 and 5) to cover a range of options acceptable to both parties.

During 1978/79 the IBA made a number of successful $2\frac{1}{2}$-channel broadcasts, using System UHJ, in conjunction with various Independent Local Radio stations. These broadcasts confirmed the ability of the system to provide a high-quality surround-sound service coupled with good coverage area. However, as experience was gained, it became increasingly doubtful whether the system was sufficiently compatible with stereo reception. This was because it caused phase shifts to be present between the two stereo channels, resulting in an imprecise quality of image. The problem was most prominent in 'pop' music programmes, where important instruments were frequently placed at the rear, but it was also noticeable in classical stage/ambience presentations. Broadly, it became apparent that these phase shifts were less acceptable on prolonged listening than earlier work (Ref. 6) would imply.

Recognising that this difficulty was sufficiently severe to stand in the way of the widespread acceptance of surround-sound broadcasting, and at the same time that

System UHJ was already close to the optimum compromise between mono, stereo and surround-sound performance obtainable with a hierarchical system, the IBA came to the view that the only way forward might be to adopt a non-hierarchical, 3-channel system (Ref. 7). At the cost of a slightly reduced service area in surround-sound reception, such a system would remove all necessity to impair the quality of stereo reception. This option would also need less expensive receiving equipment than either 2½-channel or 2-channel options. Since, however, existing proposed 3-channel systems were inadequate, the IBA proposed (Ref. 7) the development of one specifically for broadcasting.

The IBA has now developed such a 3-channel system (Ref. 8), and during 1979 used it to make a number of successful experimental broadcasts, which confirmed its feasibility. The system (known as MSC1) is defined and described later in this paper, but is not to be regarded as finally optimised; indeed, it appears likely that two or more different 3-channel matrices are really needed to give the best possible compatible stereo and surround-sound performance for such diverse material as classical stage/ambience, 'pop' music and dramatic productions (Ref. 8). System MSC1 is intended as a compromise covering all these types of material.

A REVIEW OF 2-CHANNEL, 2½-CHANNEL AND 3-CHANNEL SYSTEMS

(a) All 2-channel systems encode every horizontal direction in a different way, using only the two transmission channels of stereo. To do this with reasonable mono and stereo compatibility, they make use of broadband phase differences, as well as amplitude differences, between the channels. Surround-sound decoding is possible with a fixed decoder, ie, a time-invariant (though perhaps frequency dependent) matrix driving the loudspeakers from the two channels, but there are large phase-shifted crosstalk signals resulting in poor ('phasey') image quality, and it is doubtful whether this mode of reproduction is an improvement over good stereo. The alternative is a programme-dependent decoder, which uses the phase/amplitude information to detect the direction of the dominant sound at any moment and optimises the image quality for that sound (at the expense of weaker sounds elsewhere). Material in which important sounds are simultaneously present in different directions presents difficulties to such decoders, and becomes audible in the form of 'gain-pumping' effects. Designs for programme-dependent decoders are still under development in various quarters, ranging in cost from tens of pounds to tens of thousands of pounds; but it is doubtful whether, even at best, they will yield fully satisfactory reproduction

of serious music material, although they may be adequate for 'pop' music and some drama. Thus, 2-channel surround-sound reproduction is inherently of limited quality.

Its compatibility with stereo reception is also limited because of the inter-channel phase shifts. In all currently proposed 2-channel systems these are kept small for front-originated sounds at the expense of rear-originated sounds. Within the 'HJ' specification a centre-rear sound may be reproduced in stereo with a left/right phase difference ranging from 115° in System UHJ (Ref. 4) to 90° in the encoding favoured by the BBC (Ref. 5). The latter value, though audibly less objectionable than 115°, is still far from acceptable in stereo reception.

(b) The 2½-channel systems use a third channel of restricted bandwidth (up to 5 kHz) to convey additional audio information and are always hierarchical — ie, the two stereo channels can be decoded alone as a 2-channel surround-sound transmission. Reception of the third channel, however, enables use of a fixed decoder to give reproduction free from audible 'phasiness', and little inferior to that obtainable directly from the 3-channel B Format source. Because programme-dependent decoding is not needed, good-quality 2½-channel receiving and decoding equipment would be cheaper than the corresponding 2-channel equipment. Also, because of bandwidth restriction, 2½-channel systems have good signal/noise properties (Ref. 2).

For best results, 2½-channel systems require somewhat closer tolerances in the basic 2-channel encoding than those of 2-channel-only systems, and so suffer at least the same stereo compatibility problems. For example, within the 'HJ' specification range, only the NRDC/Ambisonics UHJ matrix (Ref. 4), with its large centre-rear phase difference of 115°, is strictly suitable for 2½-channel working.

(c) The 3-channel systems use a third channel of full bandwidth (15 kHz for FM radio). Such systems need not be hierarchical; so, two stereo channels can be chosen to give the best compatible stereo and mono performance. Thus, there need be no interchannel phase shifts. The drawback is that, in surround-sound reproduction, the signal/noise ratio is somewhat inferior to that achievable in 2½-channel working. However, there is freedom to choose a third channel giving the best compromise between signal/noise ratio and degradation of stereo reception, taking into account the spatial distribution of typical programme material. On decoding, the original B Format signals are fully recovered. The decoding equipment needed is simpler and cheaper than for 2½-channel reception.

System MSC1 (Ref. 8) was designed with the above requirements in mind.

SPECIFICATION OF SYSTEM MSC1 FOR BROADCASTING

(a) Three-channel Multiplex Transmission System

The IBA experimental transmissions (both $2\frac{1}{2}$-channel and 3-channel) have used an extension of the pilot-tone system where the instantaneous deviation is proportional to:

$$\Sigma + \Delta \sin 2\omega t + T \cos 2\omega t + 0.1 \sin \omega t$$

with $\omega = 2\pi \times 19$ kHz; and correspond to 'M' and 'S' in conventional terminology, and T is the third channel. All three channels are pre-emphasised with a 50 μs time constant.

The phase tolerance of the transmitted pilot signal needs to be tighter than for stereo, and a tolerance of at most $\pm0.5°$ is suggested in place of the $\pm3°$ given in CCIR Recommendation 450, 2–7.

Any additional pilot tones which might be used to signal the 3-channel mode are not at present defined.

(b) Matrix Encoding System MSC1

The transmitted audio signals ε, Δ, T are defined by:

$$\begin{pmatrix} \varepsilon \\ \Delta \\ T \end{pmatrix} = \begin{pmatrix} 0.9 & 0.1092 & 0 \\ 0 & 0 & 0.6897 \\ -0.5592j\lambda & 0.5592j\lambda & 0 \end{pmatrix} \begin{pmatrix} W \\ X \\ Y \end{pmatrix}$$

where $\lambda = 0.7071$; W, X, Y are the three source signals in Ambisonic B Format, defined to have relative gains of:

$$W : X : Y = 1 : \sqrt{2} \cos\theta : \sqrt{2} \sin\theta$$

respectively, where θ is the azimuth of the sound measured anticlockwise from centre-front.

A fourth transmission channel could be added to yield with-height (periphonic) reproduction, but broadcasting of this in Europe is impracticable.

The B Format signals can be obtained from coincident microphone arrays or from suitably designed pan-pots (Ref. 3). Pair-wise mixing is not recommended as a normal practice, but if required can be accommodated by using the initial transformation:

$$\begin{pmatrix} W \\ X \\ Y \end{pmatrix} = \frac{1}{2} \begin{pmatrix} K & K & K & K \\ 1 & 1 & -1 & -1 \\ 1 & -1 & 1 & -1 \end{pmatrix} \begin{pmatrix} L_F \\ R_F \\ L_B \\ R_B \end{pmatrix}$$

with $K \simeq 0.9$.

BASIC CHARACTERISTICS OF SYSTEM MSC1

The summary below follows the list of desirable characteristics given in CCIR Report 300-4 (1978), 2.2.

(a) Stereo and Mono Compatibility

System MSC1 yields a stereo signal unimpaired by the unpleasant inter-channel phase shifts which are necessarily present in 2-channel and $2\frac{1}{2}$-channel systems. Three-channel encoding permits improvement of stereo and mono compatibility up to the limits set by the differing artistic requirements of stage/ambience and 'all-round surround' material, which ideally need different encoding matrices (Ref. 8). The stereo stage width and front/back perspective chosen for System MSC1 are a compromise; but, if provision of a choice of matrices were to prove practicable, then each would still have the same general form as MSC1.

The mono capability is linked to the stereo compatibility. Rear sounds are attenuated by 3 dB relative to front sounds; this, again, is a compromise.

Two further problems of compatibility may arise through the imperfections of real stereo receivers (Refs. 8 and 9). One is that demultiplexer misalignment causes linear crosstalk between the T and Δ signals. In System MSC1 this effect is significant only for rear-originated sounds and results only in an image broadening (ie, a small interchannel phase difference) rather than an image shift. The other consists of increased distortion in some receivers when the third channel is present. With System MSC1 this problem is confined to rear-originated sounds as well as to high deviation levels; it is believed that few receivers will be noticeably affected, and that, in most cases, the audible degradation, if any, will be much less objectionable than the unavoidable phase shifts generated by the 'HJ' encoding for the same rear sounds.

(b) High Quality Surround-sound Reproduction

Three-channel systems, such as MSC1, yield reproduction equivalent to that obtainable directly from the 3-channel B Format source.

The $2\frac{1}{2}$-channel systems (eg, UHJ) yield only slightly inferior surround-sound reproduction, but 2-channel systems have been found by the IBA to be unsatisfactory even with programme-dependent decoding.

(c) Cost of Surround-sound Receivers

MSC1 3-channel reception requires a 3-channel FM tuner together with a matrix decoder which provides (after amplification) loudspeaker drive signals. Conversion of a stereo FM tuner to three channels involves either modification or replacement of the stereo decoder (demultiplexer) section, but no change before the discriminator; replacement is generally the wiser course, as the pilot-tone separation circuitry might need upgrading. A cheap high-performance 3-channel demultiplexer has been designed by the IBA.

The $2\frac{1}{2}$-channel reception follows similar lines, except

that the complexity of the matrix decoder is roughly doubled. By contrast, 2-channel reception, though needing no tuner modification, calls for a programme-dependent decoder which can be very costly and yet yield inferior reproduction. The 3-channel mode is thus the cheapest of all for reception, and will become more so if, from the start, receivers are designed as 3-channel units.

(d) Effect on Service Areas in Mono and Stereo Reception

The presence of a third channel can reduce the deviation excursion which is available to the stereo and mono signals, and hence to their service areas. In System MSC1 this effect is sharply confined to sounds encoded in a narrow sector near centre-rear. Measurements have shown (Ref. 8) that the resulting loss of headroom is less than 1 dB for sounds spread either uniformly around the listener or predominantly in front, that it reaches 2 dB for a spread of sounds across the rear quadrant, and that it reaches the theoretical limit of 4–5 dB only when a single dominant source is present at centre-rear. The behaviour is, therefore, optimised to suit the types of sound distribution which occur most frequently, and the loss will rarely exceed 1 dB.

(e) Service Area in Surround-sound Reception

The objectively calculated loss in weighted signal/noise ratio in System MSC1 surround-sound reception, compared with stereo reception of the same broadcast with the same receiver, is less than 4 dB. Subjectively, the degradation is a little worse, perhaps because noise from the rear loudspeakers is poorly masked by most signals, and is believed to be about 6 dB.

(f) Susceptibility to Interference in Surround-sound Reception

Direct measurements of this have not been made. However, since the third channel occupies the same spectrum space as the stereo difference (S or Δ) signal, the susceptibility to interfering signals, and to impulsive interference, is expected to increase (relative to stereo reception) in the same ratio as the susceptibility to random noise, ie, subjectively about 6 dB for System MSC1.

It is anticipated that this degradation may seldom occur in practice if better designs of demultiplexer are used when converting to 3-channel reception. Although it is stated (Ref. 10) that, in order to obtain the best rejection of interference and multipath distortion, there must be a 53 kHz low-pass filter after the discriminator, many receivers do not use such a filter, and consequently give inferior performance. However, there now exist circuit techniques for demultiplexing which achieve optimum performance without the expense of this filter, and these could be adopted as standard in 3-channel receivers. The IBA multiplexer (para. c above) is of this type.

(g) Effect on Frequency Assignment Plans

Transmission of the third channel does not increase the RF bandwidth needed, and should have no effect on other services.

Reception of the third channel may be about 6 dB more sensitive to interference (see above), but this difference (which does not affect the compatible stereo and mono reception) does not warrant any change in frequency assignment plans.

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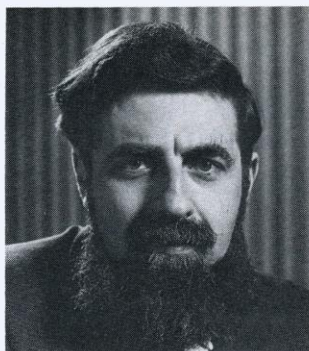
Surround-sound: An Operational Insight

by C P Daubney

Synopsis

Investigations made by the IBA into the various proposals for surround-sound systems were in two parts, theoretical and practical. This article describes how it was necessary for the IBA to make recordings of suitable test material for the practical investigations. Two vehicles were equipped as a mobile surround-sound recording control unit. Several constraints which affected the design of the equipment are explained, and a full description is given of the equipment used, including the comprehensive sound mixing desk. The associated microphone and monitoring techniques also are discussed.

INTRODUCTION

About three years ago, the IBA decided to undertake a review of the various proposals for Quadrophony which were then current; it did this, not because Quadrophony was an immediate and vital necessity for the infant Independent Local Radio (ILR) network, but because it wished to guard against international adoption of any standard which was less than optimal and which, if adopted, would be very difficult to change.

On a point of nomenclature, it soon became apparent that Quadrophony was not the right word. There is no necessary connection between the achievement of reproduced sound coming from all around the listener and the use of four, and only four, channels of information to carry the sound or of only four loudspeakers to reproduce it. In the absence of a more evocative and accurate name, the IBA decided to style it 'Surround-sound'.

In essence, the IBA review was of two parts:

(i) a theoretical one, to determine how each system provided the clues for human ears and brains to perceive the surround effect, and how well the compatible stereophonic and monophonic signals were derived;

(ii) a practical one, to see how each system fared with all the different styles of programme material and production technique.

Early in the process of making the practical review it became obvious that, with the commercial material then available, the practical part was almost impossible because no programme material, recorded by the same artists in the same environment, but in the different systems, was obtainable. As this comparison was of fundamental importance, the IBA decided to undertake the necessary recordings. A full description of this stage of the investigation is contained elsewhere in this Review. Suffice it here to say that, while a lot was achieved by using classical 'overall' microphone techniques, only very superficial work was possible in the field of popular music. This was because the facilities of the mixing equipment were very limited due to the time available for completing the work. However, the investigation would not have been complete without a detailed look at the problems of popular music recording, as well as a further look at drama, documentaries and 'classical' music. Therefore, the IBA decided to build a more extensive and permanent mobile recording facility.

This article reports on the design of the mobile unit and on the subsequent experience from the operational viewpoint. Some details of the engineering designs involved are covered elsewhere in this Review.

DESIGN PHILOSOPHY

Several constraints affected the design of the equipment:

(i) Surround-sound is a developing subject, and changes in parts of the system were likely.

(ii) The main involvement of the IBA was in conducting an investigation into systems rather than programme making.

(iii) By the nature and structure of ILR, the IBA transmitters are supplied with programmes by programme contractors; transmissions of suitable programmes, made by use of different Surround-sound techniques, are possible only with the help and co-operation of the contractors.

(iv) As the programme contractors are separate companies, each has its own philosophy and own experience of OB equipment design and layout.

(v) The time for the design and construction of the equipment, and the availability of suitable vehicles, were limited.

The keynote of the design, therefore, was flexibility.

During the early theoretical considerations, it emerged that the system known as 'Ambisonics' and which was sponsored by the British National Research Development Corporation covered the subject much more broadly than did most of its competitors; the various features were evolved from a theory of hearing, and the design covered the entire chain from microphone to loudspeaker.

As the Ambisonic design appeared to offer a system founded on a theory rather than on empirical guesswork, the IBA decided to pursue the majority of its work along the Ambisonic line.

In essence, Ambisonic technology may be summarised as follows:

For precise central decoding of Surround-sound in one (horizontal) plane, three discrete channels of information are required. (One additional channel would allow such decoding to include height.) There is no constraint on the number of microphones which can be used, nor on the number of loudspeakers, provided that these are more than three. The system is divided into four sections, each with its own format:

A-Format: Microphones — covered later in this article
B-Format: Studio Equipment
C-Format: Signal Transmission
D-Format: Decoding and Loudspeaker signals.

The sound mixing desk operates primarily in B-Format, though C-Format signals are derived in it for transmission, and the monitoring decoding is in D-Format.

In B-Format, the signals are:

W — the pressure signals from the microphones irrespective of direction

X — the front-to-back 'figure-of-eight' velocity component signals from the microphones

Y — the left-to-right 'figure-of-eight' velocity component signals from the microphones

Z — the top-to-bottom 'figure-of-eight' velocity component signals from the microphones.

The Z component is omitted for one-plane (horizontal) Surround-sound. This arrangement of signals is chosen because it is rugged and any interchannel errors manifest themselves in the least unacceptable form as symmetrical image displacements.

C-Format signals are a linear transcoding of the final mixed B-Format signals:

L — Left signal for stereophonic compatibility
R — Right signal for stereophonic compatibility
T — Third channel to allow accurate horizontal decoding
Q — Fourth channel to carry height information.

The Q component is omitted for one-plane (horizontal) Surround-sound.

In principle, D-Format signals can be derived for any number of loudspeakers greater than three, from either B or C Format inputs; the speakers are not constrained to being in a square layout. In the IBA mobile unit, four speakers are used, but decoding to six and eight speakers has been tried for certain other experiments.

With the above in mind, it was decided to use the modular 'black box' approach, i.e., to provide as many different sorts of facility in as large quantities as possible, preferably with each having individual inputs and outputs, so that the various styles of sound operation could be explored with the same equipment.

DESIGN DETAIL

Two vehicles are used; one houses the mixing and monitoring equipment and the other the multitrack recorders. The second van acts also as a cable tender. These vehicles, together with a layout of the mixing van, are illustrated in Figs. 1, 2, 3 and 4.

Single Channel Sources

Ultimately, for horizontal Surround-sound, every source has to 'acquire' the three channels of information which characterise its volume and location. Unless, as in the case of the soundfield microphone (see below), the output signal from the microphone be already in the three-channel form, all microphone sources have to be processed to derive them. The same is true of synthesised sources.

Monophonic Channels

There seems no reason to suppose that, with the advent of Surround-sound, the popular music world will suddenly change its technique of multi-microphones, etc. Therefore, the desks for such will have to contain a large

Fig. 1. The IBA Surround-sound Mobile Recording Facility is housed in these two vehicles. The nearer vehicle houses the mixing and monitoring equipment while the other provides transport for cables, etc., in addition to its main function of holding the multitrack recorders.

Fig. 2. The main items of equipment are positioned as in this plan. The layout was designed in modular form to provide a maximum variety of facilities, in as large quantities as possible, within the available space. The interconnections required for each different style of sound operation are effected on the bay jackfield.

Fig. 3. Twenty conventional monophonic channels (in two parts) about the central special three-channel control sections and the monitoring selection buttons. The penthouse contains the panning and spreading equipment, the transmission encoder and various visual and aural monitoring facilities.

Fig. 4. The experimental nature of the IBA investigations required that the programme material should be recorded in the most flexible way so that individual recordings could be used for a variety of different effects. Multitrack recorders, besides being necessary for 'pop' music recordings, served this further purpose.

number of conventional microphone channels which can amplify and control the individual sources. They must also feed multitrack tape recorders and artificial reverberation devices, feed or insert other processing devices and provide controlled line level outputs of each source for panning and mixing. In the IBA mobile, there was room for only 20 monophonic channels (and 4 monophonic groups) after space was allocated for all the other facilities. This provides sufficient channels to cope with small 'pop' sessions and is thought to be adequate to

allow exploration of the majority of different styles of programming — albeit in a small way.

The line level outputs of the monophonic sources need to be panned so that the sound appears to come from the desired direction in the soundfield. This is effected by use of special pan-pots.

Pan-Pots — Figs. 5 & 6
Essentially, a mono signal is fed into a pan-pot and the three signals (W, X & Y) are derived therein. W is
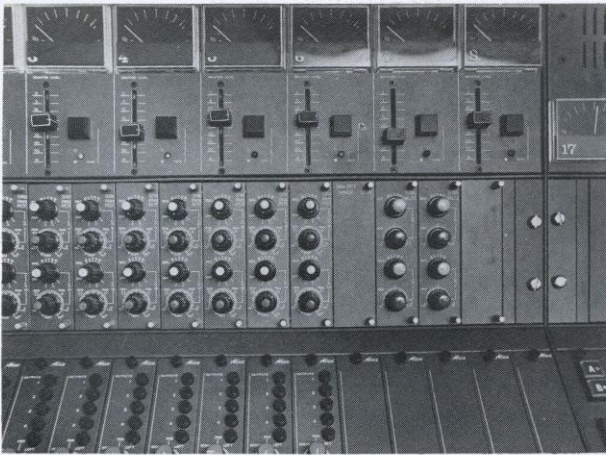
Fig. 5. Beneath a number of the multitrack PPMs are housed two of the special surround-sound facilities. To the left are shown some of the combined panning (through 360°) and spreading (diffusing) controls; each module contains two separate pan-pots and their associated pan-pot mixture selector. To the right of the pan-pots are two modules housing four Waltz controls. These allow for a previously balanced surround-sound field microphone being rotated about the listener by as much as 360°.

directly proportional to the amplitude; the relative gains and phases of X and Y provide the directional information. Thus, as the signal is panned around the circle, its position is determined by the relative values of W, X and Y. In practical terms it would be ideal to provide 360° panning on one control; in the time available, the required sine/cosine pots to achieve this could not be obtained. So, in the IBA desk, it is necessary to select on a switch the quadrant required, and then to pan on a separate control to the precise position within the quadrant. For flexibility, it would have been useful if the input and output of each pan-pot could have been available separately; however, with 20 inputs, there are 60 outputs, and the jackfield bay — Fig. 7 — was already 34 rows high.

Thus, bearing in mind that at some stage in the proceedings the 20 sets of 3 outputs had to be combined as part of the final mix, it seemed prudent to achieve some of the mixing in the pan-pot modules. As a consequence, every pan-pot can be routed to one of four pan-pot mixtures, and the outputs of the pan-pots are available only via these mixtures. However, it is possible to monitor the operation of each individual pan-pot.

When a Surround-sound pan-pot is working normally, the information contained in the three outputs W, X and Y, is sufficient to allow a proper decoding system to produce an accurately localised image. Later in this article will be found a brief discussion on the possible need for deliberately being able to delocalise an image in

a controlled way — over a limited range. This facility is available on the pan-pots.

By use of appropriate networks, it is possible to spread a panned image by a maximum of ±45° on either side of the precise position. The engineering design of this facility is described in detail elsewhere in this Review; but, in essence, the diffusion is obtained by panning different parts of the first frequency domain to different positions within the chosen amount of spread. The positions are such that a frequency sweep over the usual broadcast audio frequency range (40 Hz to 15 kHz) would cause the image to swing back and forth six times.

Spreaders — Fig. 8
Later in this article the need for delocalised sources is discussed. The achievement of such an effect is possible in the IBA desk by the use of spreaders; these are an extension of the concept of the spread on the pan-pots. In the current mono spreader design, there are no external controls, but simply one (mono) input and the three (surround) outputs. The signal swings, not merely over a maximum of ±45°, but right through 360° six times during a sweep of the normal audio frequency range.

An extension of the mono spreader is the stereo version, in which each of the two inputs is rotated in its own network, but the two are then summed to one three-channel output. At any frequency, the circuitry is arranged so that the two images corresponding to that frequency are 180° apart across the circle. This stereo version was conceived to cater for modern artificial reverberation systems which have stereo outputs.

Consideration has been given to a further improvement in the illusion, by using a four input version, but this has not yet been tried.

Three Channel Sources
Once every source has 'acquired' its three channels of information, consideration can be given to generating the final mixed three channel output.

Waltz Controls — Fig. 5
Operationally, there are many occasions when, having generated a number of three-channel sources or sub-mixtures, it is desired to rotate one or more of these sets of signals relative to another set to overlay or interleave the respective images. The Surround-sound extension of the stereo 'offset' facility is the waltz control.

One set of W, X and Y signals is fed into such controls, and from those controls is derived another set. In fact, the W signal, representing the amplitude of the signal irrespective of direction, is unaffected; the X and Y components contain the directional clues, and these are the ones which are changed as the source is rotated. In

Centre front

Front (F)

45° 45°

Left (L) 90° (R) Right

(B) Back

(a)

(b)

(c)

Maximum diffusion arc

45°

45°

(d)

Monitor feeds

Pan-pot η input

Pan-pot η

XW
XX } Pan-pot mixture 1
XY
XW
XX } Pan-pot mixture 2
XY
XW
XX } Pan-pot mixture 3
XY
XW
XX } Pan-pot mixture 4
XY

(e)
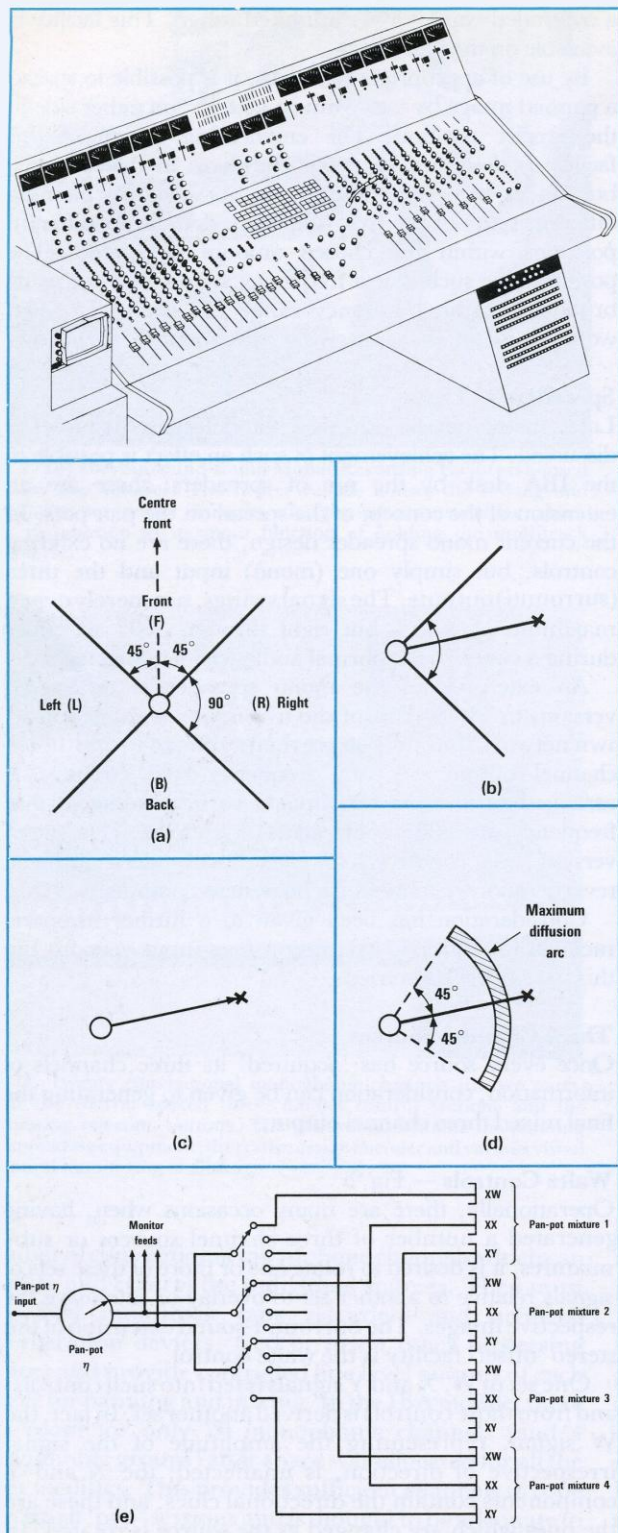
principle, this function would be provided on one control, such that one rotation of the control provided a complete rotation of the images through 360°. Because of the lack of suitable sine/cosine pots, the facility in the IBA desk is provided by choosing the number of quadrants through which it is required to rotate the images on one control, and then by 'fine tuning' the rotation on a continuously variable control. When the waltz control is set to Centre Front, there is no rotation of the images; all other shifts are derived relative to this.

Three-channel Surround Groups — Fig. 9

Surround group faders are essential to controlling the various three-channel sources used in deriving the final mix. The desk contains five such group controls, each having a three-channel input and output. A balance attenuator (in 6 dB steps from +18 dB gain to −12 dB gain) is provided on each channel, but is ganged to one control on each group, and the continuous level control is by similar means of accurately matched and ganged Voltage Controlled Attenuators (VCAs).

Since the soundfield microphone (see below) has high level outputs and contains a Z component, it is convenient to route the outputs of this microphone via one of these surround groups. In order to preserve the Z signal (when operationally convenient) at the same level as the W, X and Y signals, one of the surround groups (Group A) has four accurately matched channels.

Jackfield and Matrix — Fig. 7

Because flexibility is the keynote, as many inputs and outputs as possible are made accessible. Apart from an insertion jackfield in the desk pedestal, the great majority of these access points are provided on the main jackfield bay. Because of the need to feed tape recorders etc., as well as the main programme chains, virtually every facility has A and B outputs of every channel. Apart from the designated sources and destinations to and from the matrix, nothing is permanently connected, and the connections with the recording van also are on the jackfield.

The electrical mixing of the signals is effected partially in the pan-pot mixtures (as previously described) and then in a matrix which is housed in the jackfield bay. The designated sources and destinations of the matrix are as follows:

Fig. 6. In the IBA desk, because of a shortage in availability of the requisite components, the pan-pots are not continuously variable through 360°, but quadrant designation is used first (a); followed by the precise pan (b); once the required position has been established (c); the sound can be diffused over an angle of and below ±45° on either side (d). Finally, the output of each pan-pot is available only via one of four pan-pot mixtures (e).
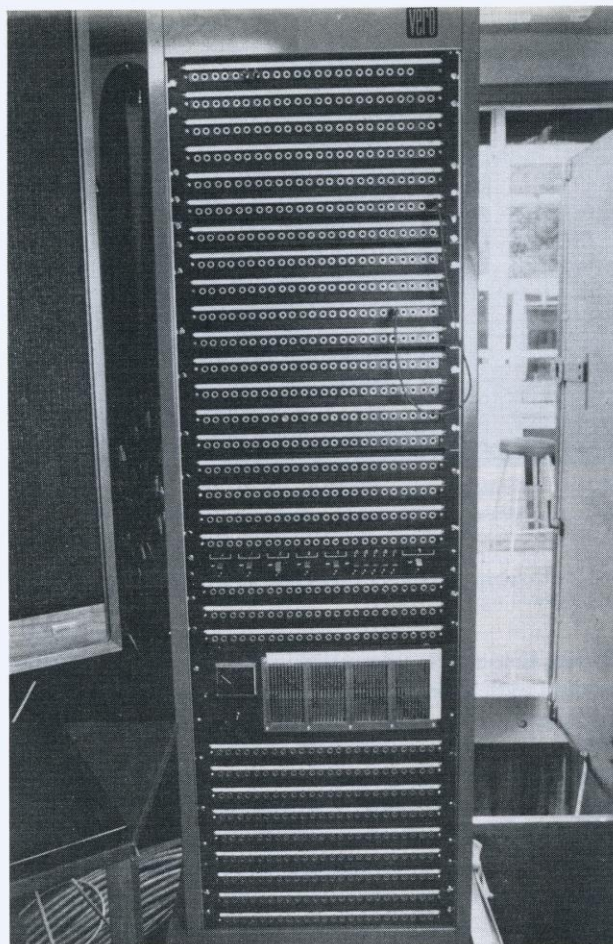
Fig. 7. Because flexibility is the keynote of an experimental recording facility, as many inputs and outputs as possible are made accessible. The vast majority of these access points are on this jackfield bay. Ultimately all the sources of sound have to be mixed to form the composite balance; the most flexible way of achieving this mixing is in a matrix such as that installed on the bay.

Fig. 8. Operationally, there is often a need to delocalise a source of sound — eg, a sound effect such as that of falling rain. The diffusion of the sound is achieved by positioning different parts of the frequency spectrum at different points around the 'circle'; a frequency sweep between 40 Hz and 15 kHz would cause the image to rotate six times around the 'circle'.

Sources			Destinations		
1. Pan-pot mix	1-W	signal	1. Surround Group	A-W	signal
2. Pan-pot mix	1-X	signal	2. Surround Group	A-X	signal
3. Pan-pot mix	1-Y	signal	3. Surround Group	A-Y	signal
4. Pan-pot mix	2-W	signal	4. Surround Group	B-W	signal
5. Pan-pot mix	2-X	signal	5. Surround Group	B-X	signal
6. Pan-pot mix	2-Y	signal	6. Surround Group	B-Y	signal
7. Pan-pot mix	3-W	signal	7. Surround Group	C-W	signal
8. Pan-pot mix	3-X	signal	8. Surround Group	C-X	signal
9. Pan-pot mix	3-Y	signal	9. Surround Group	C-Y	signal
10. Pan-pot mix	4-W	signal	10. Surround Group	D-W	signal
11. Pan-pot mix	4-X	signal	11. Surround Group	D-X	signal
12. Pan-pot mix	4-Y	signal	12. Surround Group	D-Y	signal
13. Echo (Stereo)	W	signal	13. Surround Group	E-W	signal
14. Echo (Stereo)	X	signal	14. Surround Group	E-X	signal

Sources			Destinations		
15. Echo (Stereo)	Y	signal	15. Surround Group	E-Y	signal
16. Mono Spreader	1W	signal	16. Trans Encoder	W	signal
17. Mono Spreader	1X	signal	17. Trans Encoder	X	signal
18. Mono Spreader	1Y	signal	18. Trans Encoder	Y	signal
19. Mono Spreader	2W	signal			
20. Mono Spreader	2X	signal			
21. Mono Spreader	2Y	signal			
22. Mono Spreader	3W	signal			
23. Mono Spreader	3X	signal			
24. Mono Spreader	3Y	signal			
25. Surround Group	AW	signal			
26. Surround Group	AX	signal			
27. Surround Group	AY	signal			
28. Surround Group	BW	signal			
29. Surround Group	BX	signal			
30. Surround Group	BY	signal			
31. Surround Group	CW	signal			
32. Surround Group	CX	signal			
33. Surround Group	CY	signal			
34. Surround Group	DW	signal			
35. Surround Group	DX	signal			
36. Surround Group	DY	signal			
37. Surround Group	EW	signal			
38. Surround Group	EX	signal			
39. Surround Group	EY	signal			

Fig. 9. On the right-hand side of the nearer set of monophonic channels are the special three-channel groups. These groups have balance attenuators at the top of the modules which provide equal amounts of coa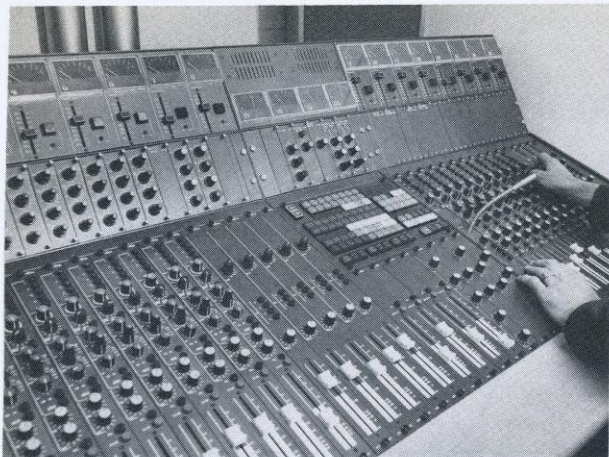rse attenuation in each of the three channels; the continuously variable control is achieved with the faders driving matched VCAs.

Fig. 11. The transcoding from the mixing format to the transmission format is effected in the module marked B-C. To the right of this is a set of decoders which allow vectorial display of the sound field on an oscilloscope and aural indication on two different sets of loudspeakers, one of which is in the vehicle. The PPMs can be switched to either B, C or D Format signals.
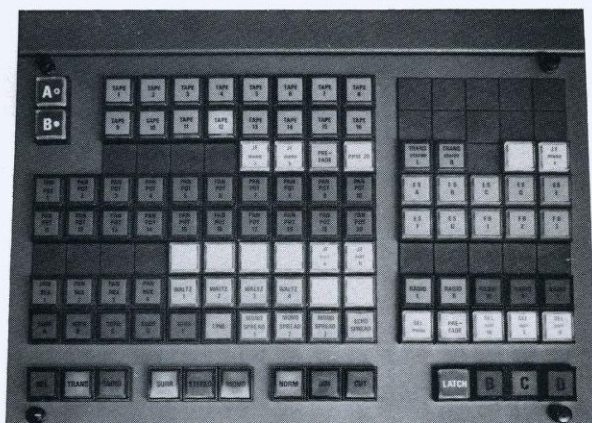

Fig. 10. In any sound control system, and particularly in an experimental one such as this, it is vital that the monitoring system shall be very comprehensive. This panel allows monitoring access to all facilities for objective assessment on the PPMs and for subjective assessment on the loudspeakers.

It is possible, therefore, to use the matrix to route signals through the surround groups in a variety of ways to suit operational requirements, and the waltz controls can be inserted in many places.

One of the destinations from the matrix is the B-to-C Format encoder.

Encoding, Decoding and Monitoring — Figs. 10 & 11
There are many ways of transcoding from B Format to C; the reasons for the particular current choice are given in another article in this Review. Expressed briefly, the B

Format final mix is transcoded in the desk and is available on the jackfield bay for line sending in the Transmission Format.

It is absolutely vital in any sound control system, and particularly in an experimental one such as this, that the monitoring is the most comprehensive possible. This makes necessary the monitoring panel on the desk (Fig. 10), and is an additional reason for the extensive jackfield facilities.

The visual monitoring facilities consist of 16 PPMs to measure either the record or replay signals from the 16-track tape recorders and PPMs 17 to 20, the feeds to which are described below. The subjective monitoring is from decoded signals fed to four high quality professional monitoring loudspeakers. PPMs 17 to 19 (Fig. 11) monitor either the B Format signals, C Format signals or, with PPM 20, the D Format signals which feed the loud-speakers. The selection of format for the PPM is made by means of the buttons in the bottom right-hand corner of the panel (Fig. 10). Unless otherwise latched, the over-riding position is C Format because that contains the Transmission Signals. PPM 20, when not monitoring one of the loudspeaker feeds, can be made to indicate the level of any of the sources on the right-hand side of the monitoring panel.

Since most of the signals to be monitored are in B Format, there is a 'B-to-D' transcoder to feed the loud-speakers; any C Format signals (Transmission output and Radio Check Off-Air input) are first converted from C-to-B. All monophonic signals are artificially encoded for monitoring as Centre Front Signals.

One of the advantages of the Ambisonic system is that the decoding arrangements are not related to either the microphone technique or the number of channels of information. For central listening, all the information necessary for accurate decoding is contained in the three channels; the decoding can be designed for any number of loudspeakers and can take account of where the speakers are positioned. Recognising that many listening rooms will not allow a completely symmetrical layout of speakers, the decoder provides Layout and Distance controls (Fig. 11) which respectively compensate for the speaker positioning and distance from the listener.

In the IBA desk, two decoders are provided, one for the vehicle itself and one for an external set of loudspeakers. The decoders are for four speakers, and the layout compensates for a 2 : 1 rectangle one way to a similar one the other way, with distance compensation (for the effect of different wavefronts) down to one metre.

INITIAL OPERATIONAL EXPERIENCE

General

After two decades of experience, engineers are still developing new techniques for stereo reproduction; hence, none could hope to do more than 'scratch the surface' of the problem of Surround-sound production in a single year. What the IBA has been able to do is to tap the considerable and diverse talent available to it through the keenness and interest of the staff of Independent Local Radio.

Engineers, Sound Balancers, Music Producers, Musicians and Hi-Fi listeners have all contributed useful comments and expertise. Much of the musical world has been examined, though considerable areas of music, documentary material and drama remain unexplored. Many lessons have been learnt, not least of which was to concentrate on simple, non-prestigious programmes in controllable and repeatable environments. By far the most useful experience has been gained from small studio sessions over which a large measure of control could be exercised — microphone positions could easily be changed, separation between sources could be controlled, and repeats were possible. The small size of the facilities put its own constraint on what was possible. However, as many 'big' sessions are only extended versions of the sort which IBA facilities would handle, there was never any point in trying a 'big' session when just as much could be learnt from a smaller one.

The first hurdle to overcome was that of learning to accept reproduced sound coming from all directions. In everyday life one accepts it without question, but artificial reproduction seemed quite different. The IBA set out to find the system which offered the perfect naturalness of everyday life — the counterpart of real high fidelity — and not merely a novel effect which would quickly become tiresome. The developing of aural recognition, and of rightly critical judgement, have also been necessary. It has been obvious, on those few occasions when everything has worked satisfactorily, that a remarkable naturalness of sound is possible. Trying to discover why those few occasions were successful, and why many others were not so, requires working analytically through what has been heard. To this end, use of the multitrack recorders and small sessions has made possible the remixing, in various ways, of the same simple material — sometimes with only very small variations — as a means of making such analyses.

It is a truism that one has to 'learn to listen' when dealing with Surround-sound. It appears that, although mono and stereo reproductions are quite different from normal everyday sounds, the human ear and brain have learned to accept them. In consequence, when presented with Surround-sound naturally reproduced, the brain registers surprise.

Perhaps the greatest apparent problem concerns the presentation of the material. One school of thought believes, presumably from an extension of what has been the norm in stereophony, that multi-microphone techniques are, to a greater or lesser extent, essential in all audio material. The other school believes that, with the ability to create around the listener that naturalness of sound which is impossible in stereo, 'classical' microphone techniques, use of a point cluster of microphones, to pick up the sound field at the wanted position, is all that is needed because the brain of the listener will do the rest.

Soundfield Microphones

'Classical' stereo microphone techniques have involved the use of a 'stereo pair' to provide the fundamental sound stage; the counterpart in Surround-sound is the soundfield microphone.

This microphone consists of a cluster of four hypercardioid microphones, approximately on the faces of a tetrahedron, and so orientated that, when projected equally onto one plane, the four capsules point (approximately) in the conventional directions of LB, LF, RF and RB. The capsules have acquired these four designations and, in the Ambisonics' terminology, this is the A Format. However, as the capsules are approximately on the faces of a tetrahedron, three-dimensional sound pick-up is possible. The capsules are spaced as closely as possible; but, in order to attempt (at least to a first order) to produce a true representation of the sound field at the centre of the tetrahedron, the amplified outputs of the capsules are equalised to take account of

their physical spacing and characteristics. At the same time, the A-to-B Format conversion is made.

The sound from the microphone is critically dependent on the capsule matching, especially as regards the precision of image localisation. One of the acid tests of any Surround-sound system, and of this design of microphone, is the 'walk around'. The evenness and naturalness of the reproduced sound from a correctly aligned array is a most useful reference against which to make other judgements.

In the classical music use of the soundfield, with the orchestra etc. conventionally arranged, the temptation to be novel and to place the listener in the midst of the orchestra has been resisted and a conventional positioning adopted. The precise positioning of the microphone relative to the orchestra depends on the style and period of the music, the size of the orchestra and the acoustics of the studio or hall — in the same way as it does in mono and stereo. Whether the derived stereo and mono is, acoustically, a less or more reverberant sound than that of normal stereo depends on the encoding system chosen for transmission.

The other matter for consideration is whether the microphone should be mounted vertically so that the sound field presented to the listener is parallel with the ground, as in everyday experience, or whether the brain is tolerant of a sloping sound field as conventional slinging of the microphone would produce. Because the four hyper-cardioid capsules are mounted on the faces of the tetrahedron and are symmetrically disposed about the horizontal plane when the microphone is in the upright position, there is a danger, with the microphone set upright near the source of sound, of the source being 'off mike' to one or other of the two capsules nearest it, and so therefore producing an imbalance in the direct sound in the derived X and Y components. As the information is not lost but contained in the unused component Z, it is possible to retrieve the situation by rotating the sound field in the vertical plane. This is easily achieved in a waltz control by overplugging the Y input with the Z component. The current version of the complete soundfield microphone which is commercially available has a steering box associated with the microphone; this steering box will produce this electrical tilt as one of its facilities.

In most broadcasts or recordings which use a 'classical' overall coverage microphone, it is frequently found necessary to supplement the output with 'spot' microphones close to certain instruments, etc. Matching the sounds from the 'spot' microphones to that of the soundfield leads to certain other possible techniques.

Other Microphone Techniques

Because the 'spot' microphones are much closer to their sources (in terms of arrival of the sound at the microphone and in terms of the sound perspective) than is the soundfield microphone, there is a danger of disparate 'lumps' of sound arising in the overall sound. Consideration of the typical distances involved shows that the sound at the output of the 'spot' microphones may be as much as 30ms ahead of that at the soundfield. With the advent of purely electronic delays, it is possible to try the effect of delaying outputs of the 'spot' microphones as to be time coincident with that from the soundfield.

Another technique which seems to provide a useful improvement (at least on initial trials) is of using the spread facility on the surround pan-pots. When mixing outputs of the 'spot' microphones with that of the soundfield, each of them will be routed through a pan-pot in order to position its output electrically and to derive the necessary three channels of information. Using a very small amount of spread on each pan-pot (not more than 5° to 10°) seems to relieve the 'lumpiness' quite considerably. Whether combined spreading and delaying would effect worthwhile improvements is a matter for future investigation.

Pan-pots and Spreaders

Apart from the use of the spread facility of the pan-pots just described, the facility is a useful way of diffusing certain instruments or vocals, particularly when an ethereal effect is required, or for spot effects in drama.

There is a likely desire to position, say, one instrument in front of another in a balance. Some initial experiments have been tried by changing the relative gains of W relative to X and Y. Certainly, when listened to at a point remote from the central position, the image can be heard to move towards the central listener's head, but the effect seems less positive when heard from that central position. The perspective of the original sound has a significant effect; and the use of spreading and delaying the sound, in addition to changing the relative gains of the three channels, would be worthy of research.

The overall spreaders have found occasional use with instruments but are very useful in coping with artificial reverberation return signals or dramatic effects such as rain and wind, all of which will probably need be delocalised completely. The design of the spreaders is critical, especially in the way the mid-band frequencies are distributed. This ensures that the dominant 'pitch' of the effects will not become 'lumpy'. Any residual 'lumpiness' can be steered to the best position by a waltz control.

Waltz Controls

It seems quite possible that, in the world of popular music, a 'motorised' version of a waltz control might soon be required, so that the soundfield can be made to spin; but

the manual control is most useful, not only for steering the output of spreaders or the 'tilting' of the soundfield microphone, but also for correcting any twist which the soundfield microphone might suffer from hanging on a cable. In addition, a soundfield microphone can be 'turned around' if, on any particular occasion, the microphone is found to sound better with an other than normal orientation. In multi-microphone operations, the waltz control is a quick and convenient means of moving one group of pre-panned instruments (say, the brass section) around the sound stage relative to another group — say, the rhythm section.

Monitoring

The greatest problem encountered so far is with the small size of the vehicle and the consequently much too small listening area. The decoding is perfectly correct only in the middle of the area; and, in the context of the vehicle, this is very small. In addition, the speakers are too close and can produce too 'intimate' an effect for some balances.

The reverberation time is well controlled and low (about $0 \cdot 2$s). When in the central listening position, the imaging seems good. Without losing too much room, this was about the highest RT which could be achieved economically. There is a school of thought which claims that a higher than conventional RT in control and listening rooms leads to improved imaging; but, within the vehicle, testing of that claim was not possible.

Operational checking of facilities is helped by the use of a vectorial display; by using the Z input of an appropriate oscilloscope in addition to the X and Y inputs, a vectorial display of the soundfield can be obtained. Pan-pots, waltz controls etc., are easily checked; some of the patterns produced during programme are very revealing — particularly when the sources of sound are in a highly reverberant building such as a cathedral. Making of the soundfield cohesive is much more difficult when working with artificially generated sound fields. With practice, the vector display can be used as a guide; but care is necessary, firstly to retain concentration on analytical listening to the balance of sound, and secondly to avoid undue interest in the vector display.

CONCLUSIONS

Just as colour television might convey greater sense of reality than does black-and-white presentation, so may Surround-sound stand superior to stereo and mono audio reception. In what might be regarded as a presentation of sound more natural, and therefore more acceptable to the human ears and brain, may lie the paradox that the listener will be the more critical of it and the more displeased when it is imperfect. When the system is used to convey an existing naturally balanced soundfield, the producer and sound balancer might need to work much harder to ensure that the realism is not distorted by the system. For those areas of material for which no pre-conceived conventions exist, it will be a matter of experience as to what extent, if any, the listening public will accept the multitude of effects which the system is able to offer.

On the assumption that the usable listening area can be made worthwhile without involving unrealistic expenditure on numbers of loudspeakers, the most important underlying question for broadcasters and recording companies is whether consumer demand will be sufficient to render economical any method of Surround-sound provision.

Time and experience in operational use will undoubtedly bring greater reality as people master the techniques, but will it be worth all the trouble and expense if the listening public is not going to want it, or be prepared to bridge the gap between real life and reproduced sound?

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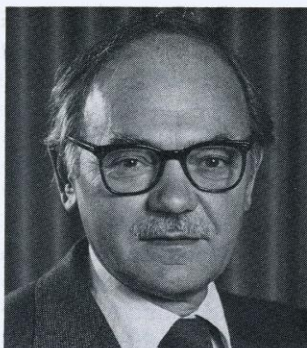
He began his career with the research laboratories of EMI Ltd, where he worked on television and microwave systems. In 1948 he joined the BBC Designs Department and in the early 1950's took a leading part in establishing television links with France — a development which led to the start of Eurovision.

He joined the former ITA in 1955. Thereafter, for several years he directed the communications activity, propagation studies and colour systems evaluation trials of the early 1960's.

In 1973 he became a Fellow of the Royal Television Society.

Future Developments in Radio

by W N Anderson

Synopsis

This article describes some of the technical radio developments which are possible within the next few years. Such items as the receiver display of station identification signals, automatic tuning of car radios, stereo broadcasting at MF, VHF receiver aerial development and broadcasting of road traffic information are discussed.

INTRODUCTION

The conclusions reached by the 1979 World Administrative Radio Conference, held in Geneva, can be taken as a guide to the main changes which are possible in the technical development of radio broadcasting throughout the next 15 years. This is not to suggest that the new frequency plans will make any significant difference to the ILR developments which are already in the pipeline, but the results will allow technical planning to proceed with some degree of confidence.

For monophonic transmission, Independent Local Radio operates on a number of channels in the MF band and offers stereophonic transmissions in the VHF band.

Programmes are broadcast simultaneously on MF and VHF. While the possibility of separating the two transmissions, for purposes of providing complementary services, has been considered, no proposal has been agreed for the implementing of such change.

The WARC had the effect of changing the frequency planning table for Region 1 in these bands, and in the following way:

MF — Increase of 1·5 kHz in the upper and lower limits, giving a range of 526·5 to 1606·5 kHz in accordance with the agreement reached at the 1975 Regional MF Planning Conference.

VHF — Addition of spectrum to 108 MHz; that is to say, utilisation of a further 8 MHz of spectrum is now internationally agreed.

While agreement for the use of extra spectrum in VHF has been accepted internationally, it is subject to regulation at the national level; and, in the UK, there seems likelihood that a period of several years will elapse before other (non-public) radio services can be taken out of this band. The assignment of frequencies to stations will be the subject of a VHF assignment conference about three years hence. It seems possible that, at that conference, the opportunity will be taken to replan VHF services using the additional spectrum to increase, substantially, the field strength in existing service areas. If this happens, it could enhance the quality of VHF reception and increase the use of VHF channels in all circumstances.

The WARC also identified the possibility of radio broadcasting from a geo-stationary satellite at frequencies of 1·0, 1·5 and 2·5 GHz; but no entries were made in the frequency table for this purpose.

LISTENING HABITS

In the seven years of ILR operation, the listening habits of the public have, to some extent, been changed. The emergence of economically sound and efficient local radio companies has been an important factor. In areas currently served by ILR 90% of adults regularly listen to radio programmes for about 23 hours each week. This

amounts to a weekly average of about 600 million listening hours. The number of people listening to ILR is more difficult to define, but surveys have shown that rather more than 50% of adults tune to an ILR station at some time during each week. There is evidence to show that the number of radio sets per household is increasing. About 97% of the population have radios in their homes, with 16% having three. It is widely known that VHF is capable of giving a quality of reception higher than that of MF. About 80% of UK homes now possess VHF equipment, and about 50% of homes are able to receive broadcasts in stereo. It might be significant that, of homes equipped with stereo and VHF receivers, the numbers in ILR areas exceed those of other areas.

The much increased popularity of radio equipment in motor vehicles has accounted for a significant part of the audience (and has brought with it a demand for local information, such as traffic reports). About 70% of cars in the UK are now equipped with radio; but only about 25% of car radios are designed to receive the VHF transmissions, and there is some evidence that, on cars equipped for VHF reception, the VHF capability is not fully used. This is probably due to the greater cost of VHF equipment and to the need to change channels rather more frequently on journeys where the vehicle crosses several service areas. It must be borne in mind that VHF service areas are more sharply defined than those of the MF service where, in many cases, the signal is receivable (at reduced quality) many miles beyond the local service areas.

FORESEEN NEEDS

Audience researches have consistently shown that many listeners have difficulty in tuning or adjusting their radio sets to the wanted transmission. In general, fixed equipment, operating from an external aerial, suffers less from this problem than does portable or mobile equipment, but many people now use portable receivers. Almost invariably any such receiver uses a rod aerial, about 600 mm in length, for VHF reception. Such an aerial is inconvenient, and could in many locations be dangerous; and there is a need for a more compact device which could be adapted to portable receivers. The characteristics of many portable receivers also render them more difficult to tune to VHF; and this, coupled with increased battery consumption, leads to a marked preference for the reception of MF services. For MF reception, the majority of portable receivers employ a ferrite rod aerial within the receiver casing.

It is known that the present receiver environment has led to a marked reluctance on the part of some listeners to change channels, because they fear they will be unable to recapture the original station. Demographic surveys have shown that this reluctance is mainly among listeners least able to adapt to the changing technology of receivers. Necessarily, this is more common among the less educated, and among the more elderly members of society. For many listeners, the reference to station identification in programme journals or announcements has little meaning when referred to the tuning scale of a receiver, and terms such as metres, kilohertz and Megahertz are not well understood. Furthermore, the calibration of receiver tuning dials has added to the confusion. The adoption of basically simple conventions, such as increasing frequency by means of tuning from left to right (or vice versa) would help the situation. However, technology is now able to offer a great deal more; and it seems that high on the list of priorities is the need to display a station identification signal on the receiver, rather than to place reliance upon listening to the programme as a means of identification.

TECHNICAL POSSIBILITIES

The introduction of new developments in radio is often inhibited by the need for compatibility with existing receiving equipment, some of which might be 20 years old. With the development of character generators and low consumption display systems, it is possible to display the station name which can be carried in coded form on a subcarrier as part of the radiated signal. The scope for the transmission of data to radio receivers is fairly good in the VHF bands because the channel bandwidth is much greater than that normally available in the MF bands. Tests have shown that, in the VHF bands, it is possible to transmit identification signals and programme related information to VHF receivers without causing interference in the programme channel.

In the USA, the VHF FM bands have been used to provide an additional channel of limited bandwidth which can offer pre-recorded music or a range of information, such as the availability of goods at local supermarkets. The extra channel is known as a Subsidiary Communications Authorisation (SCA) and is normally transmitted on a subcarrier, at a frequency of between 57 kHz and 95 kHz. More recently, it has been shown that transmission of data on specialised matters such as corn and fat stock prices, has captured an important part of the market.

In a country which is already equipped for data services, such as Teletext and Prestel, it is doubtful whether radio broadcast subcarriers would find much commercial application, except for specialist use; but a few lines of text, relevant to the broadcast programmes, might well add interest to the radio programme.

In the car radio situation, there is need for improvements related to driver safety. The diversion of an undue

amount of effort to the tuning of a radio receiver in a car can easily lead to a hazardous situation. Automatic tuning, with a search option, is an obvious candidate in this area and it seems likely that, in future, receiver manufacturers will dispense with normal mechanical tuning arrangements and substitute devices which operate by digital electronic means.

Proposals for station identification systems (Refs. 1 & 2) on FM radio have already been made by Philips in Holland and by the Swedish Telecommunications Administration. The most comprehensive proposal has been made by Sweden and is based on the transmission of an effective data rate of about 600 bits per second, on a subcarrier of 57 kHz. The actual transmission rate might need be almost twice this value, in order to provide error correction, synchronisation and address structures. The system can provide a paging service throughout the country, identification of stations, and selection of preferred types of music through appropriate digital labels attached to each programme. A search facility, on a microprocessor controlled receiver, can provide automatic selection of programme types. In addition, audio tape recorders may be automatically controlled to record programmes of given type.

In the more sophisticated types of receiver, it will be possible to display four or five words which may be used to augment the information on the programmes, such as the name of a speaker or the address for information relating to a product. The display can also cover traffic information, time, weather and public service announcements. A receiver of this complexity would need to take advantage of large scale integration of the circuits, and a fairly high volume of production, to bring the price to an acceptable level.

The WARC made no proposal in Region 1 (Europe) which might significantly affect the MF service. The present planning gives 9 kHz channel spacing and is very fully exploited, with the result that, in certain parts of the UK, there is interference during hours of darkness. On the other hand, it is the band most widely used by all listeners, and it provides the best continuity of service in car radio receivers. The sidebands of the present double sideband modulated signal are attenuated at frequencies of more than 5 kHz from the carrier, so the opportunity for introducing, above the audio modulation, signals carrying station identification information is very limited.

The stations could more easily be identified on MF, without a separate data channel, if there were fewer stations on common frequencies; and it seems possible that single sideband modulation might be a long-term solution. Proposals for the use of SSB at MF (Refs. 3 & 4) have been made. Such arrangement would allow operation of many more stations in the same frequency

band, and at a rather lower level of mutual interference than now occurs. One of the problems is that the conventional envelope detection receiver is incompatible with this type of modulation; and it is difficult to see how a transition to SSB could be arranged without serious interruption of service for the majority of listeners. Furthermore, the replacement of receivers on a national scale would be very expensive, and difficult to implement, without the promise of some new types of service. The opportunity to replan the use of the MF band is unlikely to arise for a decade, at the end of which another assignment conference will fall due.

In the present situation, it is possible that the phase modulation of the carrier could be used to derive a signalling channel (Ref. 5). This might be sufficient for station identification, but not for very much more.

STEREO BROADCASTING AT MF

The adoption, about ten years ago, of the Zenith pilot tone stereo system for use in the VHF FM bands (Ref. 6) has tended to obscure the fact that proposals for stereo transmission, using amplitude modulated signals, have been known more than 30 years. A revival of interest in this subject has occurred in the USA where there are economic arguments for providing stereo in the AM bands which are now found to be losing a major part of the audience to FM channels. In response to growing interest by the broadcasters and the electronics industry, a National AM Stereophonic Radio Committee (NAMSRC) was formed in 1975, and this body reported its findings on a number of systems which were thought to be suitable for adoption in the USA. The Federal Communications Commission is expected shortly to announce the result of a rule-making procedure (Refs. 7 & 8) on this matter.

It has been shown that stereo information can be transmitted by a number of methods within approximately the same bandwidth as a monophonic transmission. A modulation frequency limit of about 5 kHz, which is frequently used in AM broadcasting, does not greatly affect the stereo image; and, in the USA, a number of different such systems are now being tested. The transmission of stereo information is difficult to achieve at MF without harmonic distortion on the compatible monophonic signal, coupled with some increase in out-of-band radiation. Many of the systems under test accept a compromise between harmonic distortion and stereo separation, which may leave a residual distortion level of or below 5%. However, the amount of harmonic distortion depends also upon the type of IF filter in the radio receiver, and upon the difference in modulation levels in the right-hand and left-hand channels of the system. The need to signal station identification, simultaneously with

transmission of an AM stereo signal, could add further complexity to the problem of avoiding overcrowded channels.

It is too early to comment on the feasibility of introducing this technology to ILR, because the planning constraints in the UK differ from those which apply in America. However, stereo reception in the MF band might offer the average car radio listener an enhanced service, without some of the prevailing uncertainties of the VHF band.

VHF AERIAL DEVELOPMENTS

Work on electrically small aerials, at the Royal Military College of Science and at other establishments, has led to the development of a form of helical aerial which needs be only about 15% of the length of the conventional VHF rod. Some of the work has resulted from support from NRDC* and collaboration from the domestic radio industry. It is expected that the loss of signal as compared with a conventional rod aerial will be about 3 dB.

BROADCASTING OF ROAD TRAFFIC INFORMATION

An important element in the programming of the ILR stations is the transmission of information relating to road traffic conditions, both locally and nationally. This arrangement is less than ideal, because the information is transmitted to all who are tuned to the particular channel, irrespective of whether they are travellers (or intending travellers) who might need it. Also, those listeners needing the information can be long delayed before receiving the particular information they require. Various solutions to this problem are possible, but almost invariably they require special receiving equipment which can recognise a signal having a traffic information content. One system (Ref. 9), resulting from joint work between the BBC and the Road Research Laboratory in the UK, is known as CARFAX. In this arrangement, a single frequency will be used in a network of MF transmitters, giving national coverage, with the transmitters in adjacent areas operating on a time shared arrangement. This means that the traffic channel will be silent for much of the time but can be caused to interrupt another programme through a dedicated fixed tuned receiver which adds the signal to the audio section of the car radio receiver.

The second type of system is that offered by Bosch in West Germany. This is a VHF system, using a subcarrier to transmit the traffic message (Ref. 10). It has the advantage that very little additional equipment is needed at the VHF transmitting station; however, it requires in the receiver greater complexity than is needed for the BBC CARFAX system. Whether the current BBC trial of the CARFAX system will lead to a national public service is not yet known.

OTHER FACTORS INFLUENCING CHANGE IN THE UK

The past 20 years have produced many changes in the radio manufacturing industry, and many of the low-cost products such as portable radios are now imported from the Far East. This is chiefly because of disparity of labour charges. As a result, certain firms have been absorbed by larger groups. However, that part of the industry producing music centres and high quality loudspeakers appears to have expanded.

Although much of the production is now done overseas, the development facilities within the UK are responsible for design. Product planning decisions evidently need to be taken with the intention of serving very large markets; and, for this to succeed, a high degree of standardisation is desirable. In developing any new type of service, these criteria can be achieved by close collaboration between broadcasters and the receiver industry, or by industry acting alone with the intention of capturing product sales and of gaining recognition of a new standard which others will follow. Where a new feature involves a change or addition to the broadcast signal, it seems that a joint approach to the resolution of a new standard is the more likely to be effective.

The roles of the EBU and the CCIR are also important in harmonising proposals for new broadcasting systems and changes which permit the addition of new features in radio equipment. A number of questions relating to the ideas discussed in this paper have been adopted for further study.

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Aménagement Acoustique d'un Studio à Entraînement Automatique

Résumé

Cet exposé donne les grandes lignes des méthodes et de la logique de conception d'un studio de conversation faisant partie d'une station de radiodiffusion locale. Les objectifs de conception d'une zone de travail de ce genre doivent être essentiellement fonctionnels et la conformité aux normes de niveau de performance a une importance primordiale. L'ordre de présentation suit de près celui du procédé de conception dans lequel les premiers sujets à l'étude sont les besoins d'isolation contre les bruits de l'environnement existant et contre les bruits dont on prévoit l'intervention dans l'immeuble. La deuxième partie principale de l'exposé traite de l'acoustique interne du studio. Elle souligne la nécessité d'appliquer, à la conception et à la distribution des amortisseurs introduits pour contrôler la réverbération, une compréhension toute moderne de la diffusion du champ sonore.

Conception d'une Petite Station Radio

Résumé

A bien des égards, la conception d'une petite station radio ressemble à celle d'une grande. En raison des limites placées sur le personnel et sur le budget, il est important que la petite station soit conçue non seulement en fonction de l'efficacité des résultats mais aussi avec suffisamment de flexibilité pour pouvoir s'adapter à l'évolution des besoins de programmes sans grande mise de fonds supplémentaire. Les exigences de la réalisation technique dépendent du type et de la variété des programmes envisagés.

Cet article décrit la planification technique d'une station comme celle-ci, ainsi que les risques à éviter et les obligations du créateur de la station concernant l'installation.

Certains Aspects des Mesures et de L'exploitation

Résumé

L'IBA a confié aux radios locales indépendantes la responsabilité des vérifications techniques de routine pour ce qui est de l'évaluation de la qualité. Il est par conséquent essentiel que les responsables du service IBA de contrôle de la qualité travaillent en étroite collaboration avec les ingénieurs des compagnies de diffusion.

Cet article décrit les méthodes d'application des essais, conformes au catalogue des normes, effectués sur les studios des radios locales indépendantes ainsi que la raison de leur existence. L'auteur de l'article fournit une liste des principaux instruments de vérification du matériel, nécessaires pour effectuer des mesures électroniques ainsi que des informations concernant les besoins spéciaux de chaque équipement. On y discute également la possibilité d'introduire de nouveaux équipements pour améliorer la précision des vérifications et pour accélérer les procédures de vérification.

Un Réseau de Contribution pour ILR

Résumé

Les nouvelles nationales et internationales sont en permanence mises à la disposition de toutes les sociétés ILR par Independent Radio News (IRN) qui opère de concert avec London Broadcasting Company (LBC) à Londres. IRN envoie régulièrement des bulletins d'informations par l'intermédiaire d'un réseau de distribution de circuits des Postes britanniques qui s'irradient de Londres en direction de toutes les sociétés ILR. Ce réseau fonctionne depuis quelque temps déjà développé au fur et à mesure de l'entrée en service de nouvelles stations ILR.

Une nouvelle exigence s'est manifestée, et maintenant les sociétés ILR peuvent contribuer des nouvelles locales pour redistribution à quelques-unes des autres stations ILR sinon à toutes. Dans ce but, on est en train d'établir un nouveau réseau, qui sera connu sous le nom de réseau de contribution et raccordera chacune des sociétés ILR directement à IRN/LBC à Londres. Pour économiser sur les réseaux des Postes britanniques, celui-ci consistera tout d'abord en sept éperons ou branches disposés de telle façon que les sociétés se trouvant sur les éperons les plus éloignés seront raccordées par ceux qui sont le plus proche de Londres. Ainsi, toutes les sociétés autres que celles qui sont aux extrémités absolues d'un éperon auront à faire appel à la commutation pour permettre aux circuits individuels d'être raccordés en tandem selon les besoins.

Tel qu'il a été conçu, ce système de commutation utilise des tonalités à auto-acheminement qui, lorsqu'on les injecte dans le réseau au niveau de n'importe quel studio désirant contribuer une nouvelle, exciteront automatiquement tous les commutateurs situés entre ce studio et IRN/LBC, constituant ainsi un chemin continu. Les tonalités d'acheminement utilisées dans ce but sont commandés par IRN/LBC et sont distribuées aux différents studios ILR par l'intermédiaire du réseau télé-phonique national STD.

Stations Emettrices ILR de Phase II

Résumé

L'auteur passe en revue les principales caractéristiques techniques des stations émettrices ILR de la première phase et présente en les discutant quelques-uns des nouveaux concepts et équipements introduits dans la phase II à venir de la construction. Il examine les nouveaux émetteurs transistorisés à fréquence moyenne et hyperfréquence. Ceux-ci permettent des puissances d'émetteurs de 1 kW et 300 W respectivement, et il accorde une attention toute particulière au procédé de modulation utilisé par l'émetteur à fréquence moyenne. L'introduction du concept de la station 'jumelée' est analysée sous l'angle des mesures à prendre pour surmonter les problèmes de la distribution et du contrôle des programmes.

On continuera à faire appel à la polarisation circulaire pour les antennes émettrices VHF (hyperfréquence) et, en raison de l'intérêt particulier qu'évoque le sujet, l'auteur rappelle les principes de la polarisation circulaire.

Une Antenne Directive FM à Double Fréquence pour les Radios Indépendantes Locales de Londres

Résumé

Depuis 1975, l'IBA fait appel à une antenne directive FM à double fréquence pour la diffusion de ses programmes. Chaque fréquence dispose de différents diagrammes de rayonnement. Le problème le plus important a consisté à établir un zéro constant sur un arc angulaire de 301° à 316° à une fréquence donnée et d'établir un zéro similaire sur l'autre fréquence à seulement 100° de décalage par rapport à la direction du rayonnement maximum.

Cet article donne une description de la conception et de l'installation de l'antenne, de l'isolation entre émetteurs et des problèmes de réflexion.

de réception en VHF et la diffusion d'informations sur la circulation routière.

Proposition des Systèmes "Surround sound" en vue d'une Adoption en tant que Norme Internationale

Résumé

Cet article examine les systèmes à 2, 2 1/2 et 3 canaux. L'on y trouve en plus des caractéristiques fondamentales de ce système, la spécification du système de diffusion IBA 3 canaux MSC-1. L'auteur examine également l'impact de ce système sur les réseaux des services publics, les possibilités d'interférence provoquées par la réception en "Surround-sound" et le coût des récepteurs "Surround-sound".

"Surround-sound": Un Aperçu sur le Plan Opérationnel

Résumé

Les enquêtes de l'IBA sur les différentes propositions de systèmes "Surround-sound" ont été effectuées en deux parties: une partie théorique et une partie pratique. Cet article décrit comment il a été nécessaire pour l'IBA de faire des enregistrements d'essai spéciaux pour les vérifications pratiques. Deux véhicules ont été équipés comme unités de contrôle mobiles des enregistrements "Surround-sound". L'article donne également plusieurs raisons qui ont affecté la conception de cet équipement ainsi qu'une description complète de l'équipement utilisé, présentant le pupitre de mixage du son dans son intégralité. On y trouvera également débattues les techniques connexes de prise du son au microphone et de contrôle de la prise du son.

Développements Futurs de la Radio

Résumé

Cet article présente un certain nombre de développements techniques qui seront possibles dans les années à venir. On y traite, entre autres, l'affichage des signaux d'identification d'une station sélectionnée sur un récepteur, les possibilités d'accord automatique des auto-radios, les diffusions stéréophoniques sur Modulation de Fréquence, le développement des antennes

Akustisches Design eines Selbstantriebstudios

Übersicht

In diesem Artikel werden Methoden und Logik für das Design eines zu einer örtlichen Rundfunkstation gehörenden Sprachstudios vom Verfasser dargelegt. Dem Design eines derartigen Arbeitsbereiches müssen in erster Linie funktionelle Erwägungen zugrunde liegen, und die Erfüllung der Leistungsspezifikationen ist von größter Wichtigkeit. Die Art der Präsentation geschieht in enger Anlehnung an die Designmethode, bei der die Anforderungen der Isolierung gegen vorhandene Umweltsgeräusche und gegen die erwartungsgemäß innerhalb des Gebäudes entstehenden Störungen an erster Stelle zu berücksichtigen sind. Der zweite Hauptteil des Artikels befaßt sich mit der internen Akustik des Studios. Darin wird betont, bei der Entwicklung und Verteilung der zur Widerhallkontrolle aufgestellten Absorber die neuesten Kenntnisse hinsichtlich der Diffusion des Tonfeldes anzuwenden.

Design einer kleinen Radiostation

Übersicht

In vielerlei Hinsicht ist die Auslegung einer kleinen Radiostation der einer größeren ähnlich. Die Einschränkungen in bezug auf Personal und Kosten erfordern, daß das Design einer kleinen Station nicht nur zweckdienlich, sondern auch entsprechend flexibel sein muß, um sich ohne zusätzliche übermäßige Ausgaben den Veränderungen bei den Programmerfordernissen anzupassen. Die technischen Designerfordernisse richten sich nach der Art und Unterschiedlichkeit der geplanten Programme.

In diesem Artikel werden die technische Planung einer derartigen Station, gewisse zu vermeidende Risiken sowie die Aufgaben des Stationsdesigners an Ort und Stelle beschrieben.

Aspekte der Messung und des Betriebs

Übersicht

IBA hat die Verantwortung für routinemäßig durchzuführende technische Qualitätsbewertungen den unabhängigen örtlichen Rundfunkstationen übertragen. Es ist daher für die IBA Qualitätskontrollabteilung wichtig, mit den Programmgestaltungsingenieuren eine enge Verbindung zu unterhalten.

Dieser Artikel beschreibt die Gründe und Methoden, die sich für die Anwendung von Übungstestkodes in ILR Studiozentren ergeben. Der Verfasser legt eine Liste der Hauptgeräte der Testausrüstung vor, die zur elektronischen Messung erforderlich sind, zusammen mit Anmerkungen bezüglich der Sonderanforderungen der einzelnen Ausrüstungen. Die Möglichkeit der Einführung neuer Geräte zur Verbesserung der Bewertungsgenauigkeit und zur Beschleunigung der Testverfahren wird hierbei erläutert.

Ein Beitragsnetz für ILR

Übersicht

Nationale und internationale Nachrichten sind stets von Independent Radio News (IRN), das mit der London Broadcasting Company (LBC) zusammenarbeitet, erhältlich. Mittels eines Verteilernetzes von Postamtkreisen, das von London nach außen an sämtliche ILR Gesellschaften ausstrahlt, wird von IRN regelmäßig ein Nachrichtenbulletin ausgesandt. Dieses Netz ist schon seit einiger Zeit in Betrieb und wird entsprechend der Inbetriebnahme zusätzlicher ILR Stationen erweitert werden.

Eine weitere Forderung besteht darin, daß ILR Gesellschaften Mitteilungen lokaler Ereignisse zur Wiederverteilung an einige bzw. alle anderen ILR Stationen als Beitrag übersenden. Aus diesem Grunde wird z.Zt. ein neues Netz, das als das Beitragsnetz bekannt ist, eingerichtet, und zwar zur direkten Verbindung jeder ILR Gesellschaft mit IRN/LBC in London. Um die Postamtkreise wirtschaftlicher zu gestalten, wird das Netz zunächst aus sieben Spuren oder Zweigen bestehen, die so angeordnet sind, daß die entferntliegenderen Gesellschaften auf jeder Spur über diejenigen, die London näher liegen, verbunden werden. Für alle Gesellschaften — außer denen, die sich an den äußersten Enden einer Spur befinden — ist daher eine Schaltung erforderlich, damit nach Bedarf die einzelnen Kreise hintereinander verbunden werden können.

Das entwickelte Schaltsystem benutzt Töne mit Eigenleitweglenkung, die — wenn sie von irgendeinem Studio, das eine Nachricht übermitteln möchte, eingespeist werden — automatisch sämtliche Schalter zwischen dem Studio und IRN/LBC betätigt und somit eine kontinuierliche Bahn erzeugt. Die zu diesem Zweck benutzten Töne mit Leitweglenkung stehen unter der Kontrolle von IRN/LBC und werden über das nationale Selbstwähler-ferndienstnetz an die verschiedenen ILR Studios verteilt.

ILR Sendestationen der Phase II

Übersicht

Der Verfasser gibt einen Überblick der technischen Hauptmerkmale für die erste Phase der ILR Sendestationen und erörtert einige der neuen Konzepte und Ausrüstungen, die in der zukünftigen Bauphase II eingeführt werden. Die neuen Mittelfrequenz- und UKW-Sender werden besprochen. Diese geben eine Senderleistung von maximal 1 kW und 300 W entsprechend, und es wird auf das von dem Mittelfrequenzsender benutzte Modulationsverfahren besonders hingewiesen. Die Einführung des ''verzwillingten'' Stationskonzepts wird besprochen, und zwar in bezug auf die Maßnahmen zur Ausschaltung der Probleme der Programmverteilung und -überwachung.

Zirkularpolarisation wird weiterhin für die UKW-Sendeantennen benutzt werden, und aufgrund des für dieses Gebiet gezeigten besonderen Interesses wird ein Überblick über die Prinzipien der Zirkularpolarisation gegeben.

Eine MF Zweifrequenz-Peilantenne für ILR London

Übersicht

Die IBA hat seit 1975 im Programmdienst ein MF Zweifrequenz-Peilantennensystem in Betrieb. Für jede Frequenz sind unterschiedliche Antennencharakteristiken vorgesehen. Das wesentlichste Problem war das Herstellen einer konstanten Nullstelle über einem Lichtbogenwinkel von 301° bis 316° auf der einen Frequenz und einer ähnlichen Nullstelle, die nur 100° von der Richtung der maximalen Strahlung abweicht, auf der anderen Frequenz. Der vorliegende Artikel gibt eine Beschreibung des Antennendesigns und der Anlage sowie der Trennung zwischen den Sendern und der Rückstrahlungsprobleme.

"Surround-sound" — Systeme, die zur Annahme als Internationaler Standard Vorgeschlagen Werden

Übersicht

Dieser Artikel gibt eine Übersicht der 2-Kanal-, 2½-Kanal- und 3-Kanal-Systeme. Die Spezifikationen des IBA 3-Kanal-Übertragungssystems MSC-1 wird hier zusammen mit den Grundmerkmalen des Systems erläutert. Der Verfasser erörtert die Auswirkung des Systems auf den Übertragungsdienstbereich, die Empfindlichkeit gegenüber Interferenz von "Surround-sound"-Empfang sowie die Kosten der "Surround-sound"-Empfänger.

"Surround-sound": eine Betriebseinsicht

Übersicht

Die von IBA in bezug auf die verschiedenen Vorschläge für "Surround-sound"-Systeme angestellten Untersuchungen untergliederten sich in den theoretischen und praktischen Teil. Der vorliegende Artikel beschreibt, wieso es für IBA erforderlich war, für die praktischen Untersuchungen von geeignetem Testmaterial Aufzeichnungen vorzunehmen. Zwei Fahrzeuge wurden als mobile Kontrolleinheit zur "Surround-sound" Aufzeichnung ausgestattet. Verschiedene Beschränkungen, die das Design der Ausrüstung beeinträchtigen, werden erläutert, und es wird eine ausführliche Beschreibung der benutzten Geräte, einschließliche des kompletten Tonmischpults, gegeben. Die dazugehörigen Mikrophon- und Überwachungsverfahren werden ebenfalls besprochen.

Zukünftige Radioentwicklungen

Übersicht

Dieser Artikel enthält eine Beschreibung einiger innerhalb der nächsten Jahre möglichen technischen Neuentwicklungen. Zu den erörterten Punkten gehören die Empfängeranzeige von Senderkennsignalen, das automatische Einstellen von Autoradios, Stereo-Übertragung auf Mittelfrequenz, Antennenentwicklung für UKW-Empfänger und das Übertragen von Straßenverkehrs-informationen.

El Diseño Acústico de un Estudio Autoaccionado

Sinopsis

Este informe resume los métodos del autor y el fundamento del diseño de un estudio acústico que forma parte de una estación local de radiodifusión. Los objetivos de diseño de dicha área de trabajo deberán ser primeramente funcionales y es de suma importancia satisfacer las especificaciones de rendimiento. El orden de presentación sigue de cerca el proceso de diseño, en el cual los requisitos de aislamiento contra el ruido ambiental existente y contra el ruido a producirse dentro del edificio, deberán considerarse en primer término. La segunda sección de este informe trata sobre la acústica interna del estudio. Destaca la necesidad de aplicar el conocimiento moderno de la difusión del campo acústico al diseño y distribución de los amortiguadores introducidos para controlar la reverberación.

Diseño de una Estación Pequeña de Radio

Sinopsis

En muchos aspectos el diseño de una estación pequeña de radio es similar al de una de mayor tamaño. Las restricciones de personal y presupuesto exigen que el diseño de una estación pequeña sea correcto y lo suficientemente flexible como para adaptarse a los cambios en los requisitos de programas sin causar gastos excesivos adicionales. Los requisitos técnicos del diseño dependen del tipo y variedad de los programas planificados.

Este artículo describe la planificación técnica de dicha estación, los peligros que deberán evitarse, y los deberes in situ del proyectista de la estación.

Aspectos de Medición y Operación

Sinopsis

La IBA ha delegado la responsabilidad de evaluar en forma rutinaria la calidad técnica a las compañías radiofónicas locales independientes. Por lo tanto, es esencial que la sección de control de la calidad de la IBA mantenga un contacto estrecho con los ingenieros de programas de las compañías.

Este artículo describe las razones y métodos de aplicar pruebas de Códigos de Práctica en los centros de estudios de ILR. El autor suministra una lista de los principales items de equipo de prueba necesarios para llevar a cabo las mediciones electrónicas junto con notas relacionadas con los requisitos especiales de cada equipo. Se discute la posibilidad de introducir nuevo equipo para mejorar la exactitud de las evaluaciones y llevar a cabo con prontitud los procedimientos de prueba.

Una Red de Contribución para ILR

Sinopsis

Las noticias nacionales e internacionales están constantemente disponibles para todas las compañías de ILR de Independent Radio News (IRN) que opera en colaboración con la London Broadcasting Company (LBC) (Compañía Radiodifusora de Londres) en dicha ciudad. Los boletines de noticias regulares se envían desde IRN por medio de una red de distribución de circuitos de Correos y Telégrafos que radian desde Londres a todas las compañías de ILR. Esta red ha estado en operación desde hace algún tiempo y se extenderá a medida que se pongan en servicio otras estaciones de ILR.

Se ha originado un requisito adicional por medio del que las compañías de ILR pueden contribuir con noticias de acontecimientos locales para su redistribución a algunas o todas las demás estaciones de ILR. Para este fin, se está instalando actualmente una nueva red, denominada la red de contribución, para enlazar cada compañía de ILR con IRN/LBC en Londres. A fin de economizar en los circuitos de Correos y Telégrafos, consistirá inicialmente en siete derivaciones o ramificaciones, dispuestas de modo que las compañías más alejadas en cada derivación se enlacen a través de las que se encuentren más cerca de Londres. Por lo tanto, será necesario un sistema de conmutación en todas las compañías que no se encuentren en los extremos terminales de una derivación para poder conectar los circuitos individuales en serie según fuere requerido.

El sistema de conmutación diseñado aprovecha los tonos autodirigibles que, al ser inyectados en la red en cualquier estudio que requiera contribuir con un ítem de noticias, activarán automáticamente todos los conmutadores entre dicho estudio y IRN/LBC, para suministrar una línea continua. Los tonos dirigibles utilizados para este fin se encuentran bajo el control de IRN/LBC y se distribuyen a los distintos estudios de ILR a través de la red telefónica nacional STD.

Fase II de las Estaciones Transmisoras ILR

Sinopsis

El autor revisa las características técnicas más importantes de la primera fase de las estaciones transmisoras ILR y discute algunos de los nuevos conceptos y equipos introducidos en la próxima fase II de construcción. Se discute sobre los nuevos transmisores de componentes sólidos MF y VHF. Estos suministran energías de transmisión de 1 kW y 300 W respectivamente, y se concede especial atención al proceso de modulación utilizado por el transmisor de MF. Se discute la introducción del concepto de estación ''gemela'' en términos de las medidas a tomar para superar los problemas de distribución y control de programas.

Continuará utilizándose la polarización circular para las antenas transmisoras de VHF y, debido al especial interés demostrado en este área, se ha incluido una revisión de los principios de la polarización circular.

Antena Direccional FM de Frecuencia Doble Para ILR (Radios Locales Independientes) de Londres

Sinopsis

IBA (Autoridades de Radiodifusión Independiente) opera desde 1975 una antena direccional FM de frecuencia doble en el servicio de programas. Se suministran modelos diferentes de reflexión en cada frecuencia. El problema más significativo fue el de establecer una reflexión nula constante sobre un arco angular de 301° a 316° en una frecuencia y una reflexión nula similar de sólo 100° fuera de la dirección de reflexión máxima en la otra frecuencia. Este artículo describe el diseño e instalación de la antena, separación entre los transmisores y problemas de rerreflexión.

Sistemas ''Surround-sound'' Propuestos para Adoptarse como Norma Internacional

Sinopsis

Este artículo suministra una revisión de los sistemas de 2 canales, 2½ canales y 3 canales. Se suministra la especificación del sistema MSC-1 de 3 canales de IBA para transmisión junto con las características básicas del sistema. El autor discute sobre el efecto del sistema en las áreas de servicio, susceptibilidad a

interferencia de la recepción ''Surround-sound'', y el coste de los receptores de sonido.

''Surround-sound'': Un Examen desde el Punto de Vista del Funcionamiento

Sinopsis

Las investigaciones llevadas a cabo por la IBA sobre las distintas propuestas para los sistemas ''Surround-sound'' se dividen en dos partes, teóricas y prácticas. Este artículo describe la necesidad de la IBA de realizar grabaciones de material de prueba adecuado para las investigaciones prácticas. Se equiparon dos vehículos como una unidad de control de grabación ''Surround-sound'' móvil. Se explican varias restricciones que afectaron el diseño del equipo y se brinda una descripción completa del equipo utilizado, incluyendo el completo tablero mezclador de sonidos. También se discute sobre las técnicas de control y micrófono relacionadas.

Desarrollos Futuros en Radiodifusión

Sinopsis

Este artículo describe algunos de los desarrollos técnicos que serán posibles dentro de los próximos años. Se discutirán temas como presentación visual del receptor de las señales de identificación de la estación, sintonización automática de las radios de automóviles, transmisión estereofónica en FM, desarrollo de la antena para el receptor VHF y transmisión de información sobre tráfico de carretera.

IBA TECHNICAL REVIEW

* Out of Print